HOUYUAN® IP

PBX-02\04\08 Product

Guide

Version: 2.0

2012

Contact HOUYUAN

The Introduction of HOUYUAN

HOUYUAN Technologies is a global leader providing next-generation converged communication products and services to Small and Medium-Sized Enterprises ("SMEs") and service providers. Our flagship IP PBX® Series products seamlessly integrate voice, data, security, IT applications and real-time collaboration. Our converged service platforms for enterprises create long-term value for our customers by increasing revenue opportunities, enhancing communication efficiency and reducing operational costs.

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Content

1.0 Introduction of IP PBX-02\04\08

1.1 IPPBX-02\04\08

The IP PBX-02\04\08 is a complete Asterisk Appliance with combination of FXO/FXS channels. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT functions. It provides a solid, uniform platform for Mobile and VoIP communications. Targeting for SOHO user and SMB market with an easy to use graphical interface, HOUYUAN IP PBX provides a cost-saving solution on their telecommunication/data needs. With these devices, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet, FXO/FXS and PSTN network.

1.2 Hardware

CPU: 400MHz Blackfin 532 Chip
2 x FXO/FXS ports and four analog ports
NAND flash 256 M SDRAM 64M

1.3 System

Open Source uClinux

1.4 Features

FXO/FXS, ISDN
Support g711/g729 codec
Voicemail
Voicemail groups
3-way Calling
Conferencing
Follow Me
Call Feature
In directory
Call Waiting
Call Queues
Pickup
Group Ring
Group

Is Agent
Music On Hold
Voice Menus
Voice menus Prompts
Time
intervals
Backup
Update

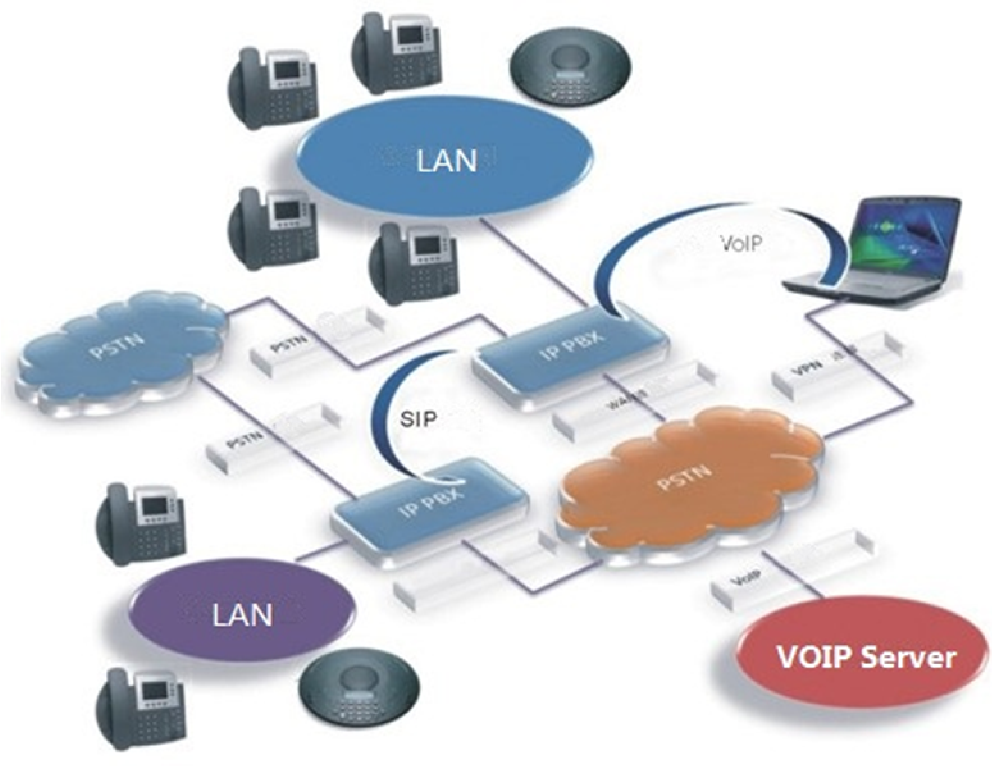
1.5 Applications

SOHO/SMB telephony system
Hosted service
IVR system

1.6 Interface

1 X RJ45 port.
1 X Power port.
1 X RS232 port.
8 X FXO/FXS channels.

Overview of the IP PBX-08



2.0 Access to the IP PBX-02\04\08

2.1 HOW to Login

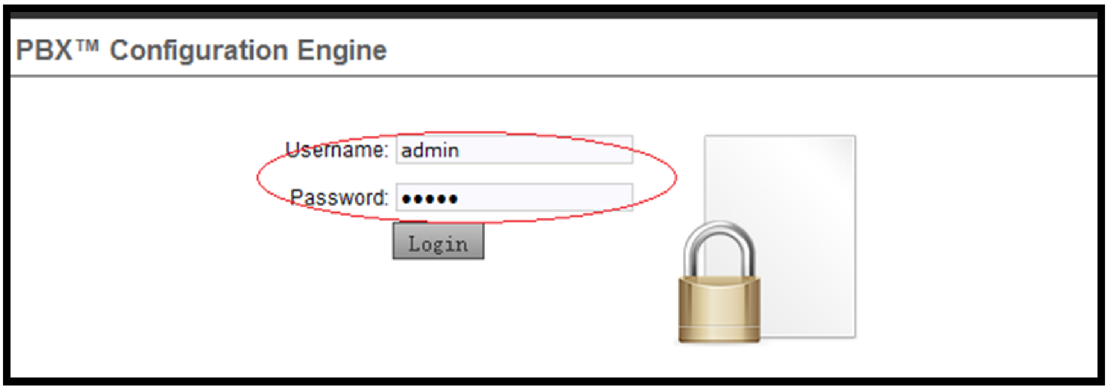
You need a PC to access to the IP PBX-02\04\08, there are four ways for you to access the IP PBX-02\04\08:

1. Web page
2. SSH
3. Console port access by RS232 console cable

In order to access to IP PBX-02\04\08 by the first three ways, Users have to check that if your network connection between IP PBX-02\04\08 and PC is OK. If it does not connect between IP PBX-02\04\08 and PC, users can try to use the last way to access to IP PBX-02\04\08 and change the IP address for IP PBX-02\04\08.

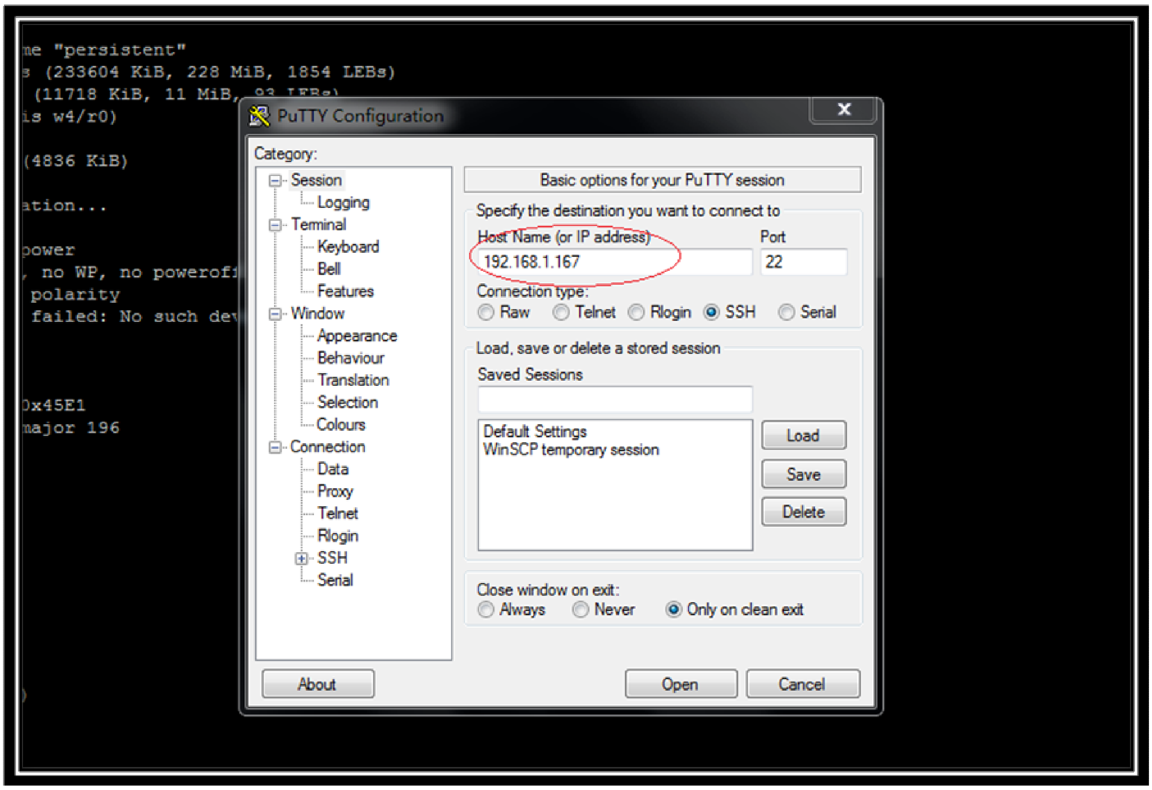
2.11 Web

WEB URL : 192.168.1.167
Username: admin
Default Password: admin



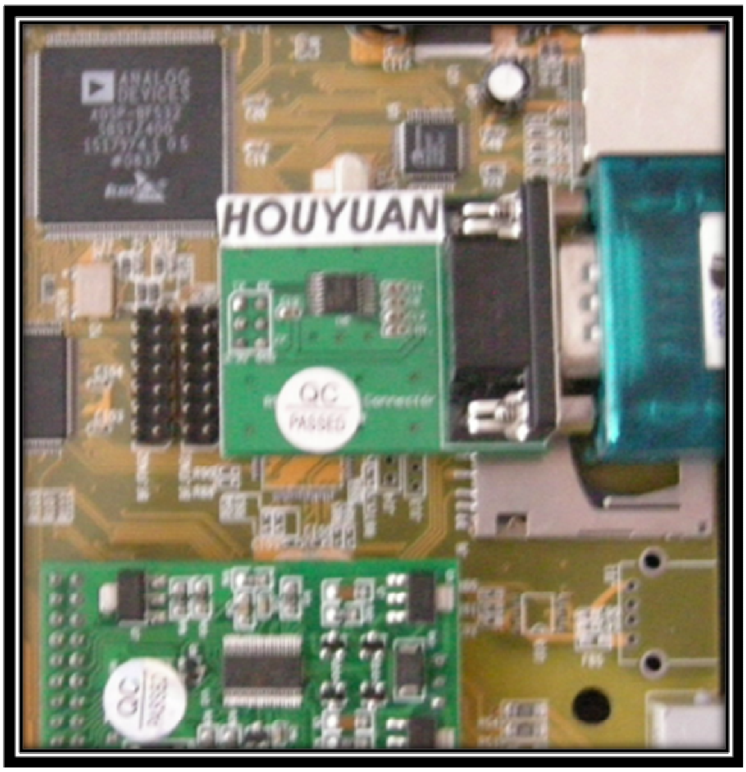
2.12 SSh

WEB URL : 192.168.1.167
Username: admin
Default Password: uClinux



2.13 RS232(Console Post or minicom)

1. Connect the console port of IP PBX-02\04\08 to your PC's console port with RS232 console cable.



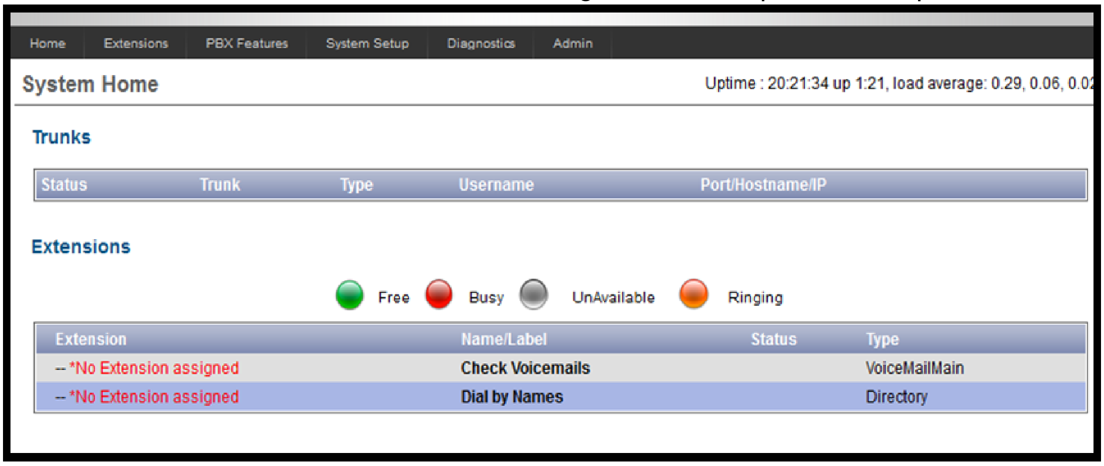
2. Run the HyperTerminal, and set up the console port like the following: Bits per second: 115200
   Data bits : 8
   Parity: None
   Stop bits: 1
   Flow control: None

```
lqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x A -    Serial Device      : /dev/ttyUSB0                                x
x B - Lockfile Location     : /var/lock                                   x
x C -   Callin Program      :                                             x
x D -  Callout Program      :                                             x
x E -    Bps/Par/Bits       : 115200 8N1                                  x
x F - Hardware Flow Control : No                                          x
x G - Software Flow Control : No                                          x
x                                                                         x
x    Change which setting?                                                x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
        x Screen and keyboard      x
        x Save setup as dfl        x
        x Save setup as..          x
        x Exit                     x
        x Exit from Minicom        x
        mqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

3.0 Web Operation of IPPBX-02\04\08

3.1 Home

In the system status screen, it displays the functions users configured, such as: trunks, extensions, conference and so on. The following table is the options description of trunks.



Status	Trunk	Type	Username	Port/Hostname/IP

Extension	Name/Label	Status	Type
-- *No Extension assigned	Check Voicemails		VoiceMailMain
-- *No Extension assigned	Dial by Names		Directory

Name	Description
Status	The register status of trunks
Trunk	The name of trunks
Type	The type of trunks
Username	The username of SIP/IAX trunk
Port/Hostname/	IP Address/port

1.The register status of trunks include three kinds: Unregistered, Request Sent, Registered.
2.The type of trunks : VoIP trunk including SIP and IAX; Analog trunk; Service Provider.

The parameter of extensions in the following table:

Name	Description
Extension	The status of users
Name/label	The name of users
Status	Display voice message
Type	SIP users/IAX users/Analog users

1. There are four kinds status of users, when the light of “Extension” list displays gray , means the user does not register that is Unavailable; when the light of “Extension” list displays green , means the user is Free; when the light of “Extension” list displays orange , means the user is Ringing; when the light of “Extension” list displays red , means the user is Busy.
2. Status: This parameter displays if other users leave messages, Messages : 0/0, the figure front of “ / ” displays the new messages amount; the figure behind of displays the old messages amount.

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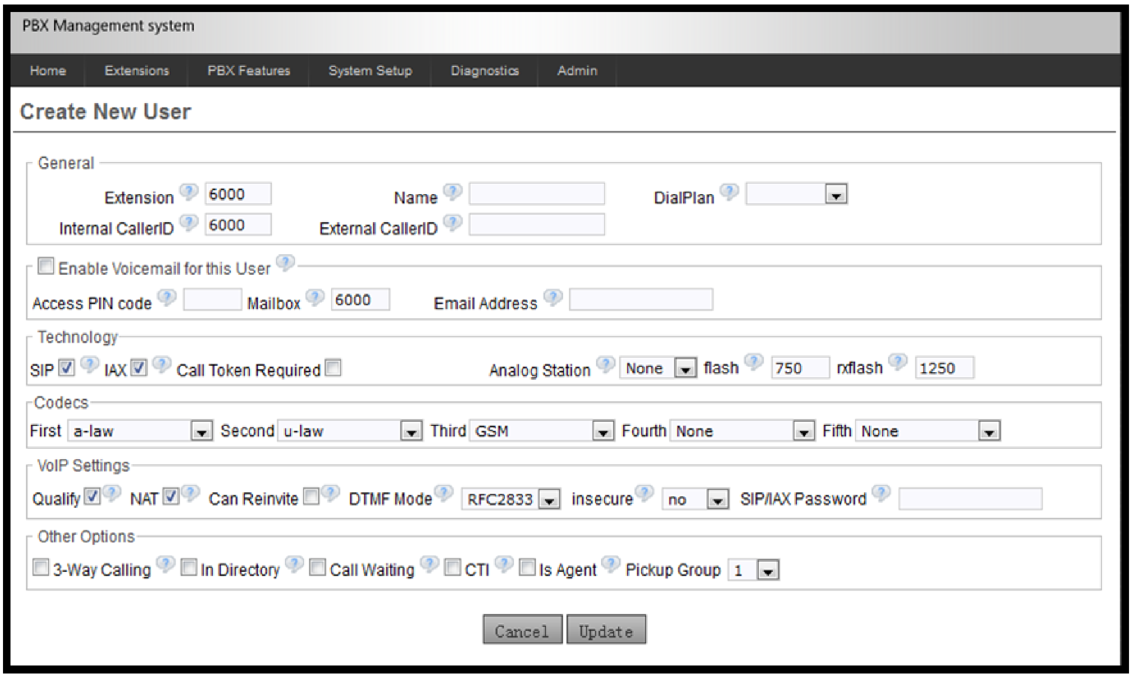
3.2 Extensions



Users component is used to add or remove Analog, SIP, IAX extension.

Click on Create New User button in the web of IP PBX-02\04\08, users can create SIP/IAX User and Analog

Name	Description	Type	Default
Extension	The numbered extension	Textbox	6001
Name	A character-based name for this user	Textbox	Null
DialPlan	DialPlans are sets of calling rules and can be managed	ComboBox	Null
CallerID	The Caller ID (CID) string used when this user calls another internal user.	Textbox	Null
OutBound CallerID	Caller ID that would be applied for out bound calls from this user. Note that your ability to manipulate your outbound Caller ID may be limited by your VoIP	Textbox	Null
Enable	Check this box if the user should have a voicemail	Selected	Not
Voicemail for	account		selected
VoiceMail	Voicemail Password for this user	Textbox	Null
Mailbox	Voicemail Mailbox for this user	Textbox	Null
Email Address	The e-mail address for this user	Textbox	Null
SIP	Check this option if the User or Phone is using SIP or is a SIP device	selected	selected
IAX	Check this option if the User or Phone is using IAX or is an IAX device	selected	selected

Analog Station	If this user is attached to an analog port on the system, please choose the port number here	ComboBox	Null
Codec	Choose priority codec	ComboBo	u-law/GS
NAT	Try this setting when Asterisk is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP	selected	selected
Can Reinvite	By default, Asterisk will route the media steams from SIP endpoints through itself. Enabling this option causes asterisk to attempt to negotiate the endpoints to route the media stream directly, bypassing asterisk. It is not always	selected	Not selected
DTMF Mode	Set default dtmfmode for sending DTMF. info : SIP INFO messages;inband : Inband audio (requires 64 kbit codec -alaw, ulaw); auto : Use rfc2833 if offered,	ComboBo x	rfc2833
3-Way Calling	Check this option if the User or Phone should have 3-Way Calling capability.	selected	Not select
In Directory	Check this option if the user is to be listed in the system telephone directory.	selected	Not select
Call Waiting	Check this option if the User or Phone should have Call-Waiting capability	selected	Not select
Is Agent	Check this option if this User or Phone is a Call Queue Member (Agent)	selected	Not selected
Pickup Group	If a user called A and another user called B in the same group,A can pick up the phone taking the place of	selected	Not selected

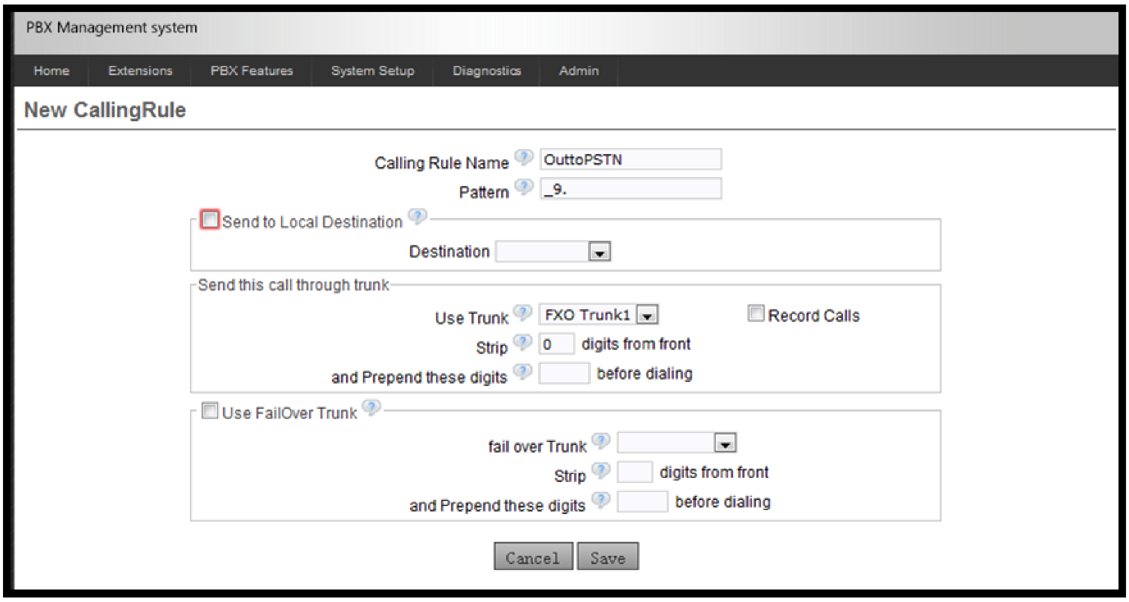
1. Analog Station: When users want to create Analog Users, please choose the FXS ports.
2. Codec Preference: Support g711u-law/g711a-law/g729/FXO/FXS

3. Attension: in the textbox of Extension, the value users set is limited to a range, they can adjust the range in the Options option to meet their requirement.

3.3 PBX features

3.311 Outgoing Calling Rule

Outgoing calling rules is used to route an outgoing call, when users make an external call, which trunk and what dial-pattern the call used are configured in outgoing calling rules. Please select the Outgoing Calling Rules option, then Click on New Calling Rule button, the parameters of the Outgoing Calling Rules are in the following table:



Name	Description	Type	Default
Calling Rule Name	The name of the Calling rule	Textbox	Null
Pattern	The dialing rule	Textbox	Null
Send to Local Destination	If this option is checked and Destination is defined, calls matching the specified pattern	selected	no select
Destination	Choose the Local Destionation:User/VoiceMenu/Hungup…	ComboBox	Null
Use trunk	Defines the Trunk that calls, matching the specified pattern, will be placed through.	ComboBox	Null
Strip	Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the	Textbox	Null
Prepend these digits	Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog	Textbox	Null
Use Failover Trunk	Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down If "Use Failover Trunk" is checked and "Failover trunk" is defined, then calls that cannot be placed via the regular trunk may have a secondary trunk defined. If a user's primary trunk is a VoIP trunk, but one wants calls to use the PSTN when the VoIP trunk isn't available, this option	selected	no select
Fail over trunk	Choose the trunk	ComboBox	ComboBox

Pattern: X ... Any Digit from 0-9; Z ... Any Digit from 1-9; N ... Any Digit from 2-9; [12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9); .... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself); ! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible. For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.

Strip: Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in Use Trunk. For example, want users to dial 9 before their long distance calls; however one does not dial 9 before those callsre placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.

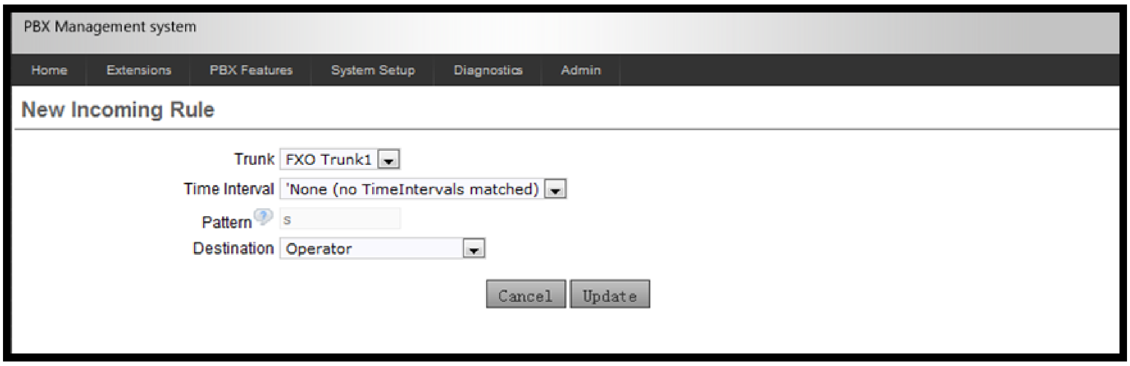
The way of outgoing calling

Every time you dial a number, asterisk will do the following in strict order:

- Examine the number you dialed.
- Compare the number with the pattern that you have defined in your first outgoing rule and if matches, it will initiate the call using that trunk. If it does not match, it will compare the number with the pattern that you have defined in the second outgoing rule and so on.
- Pass the number to the appropriate trunk to make the call.

3.312 Incoming Calling Rule

This is where the behavior of incoming calls from all trunks is being handled. When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, digital receptionist, voice menu or queue. For this purpose, Incoming Calling Rules need to be set up.

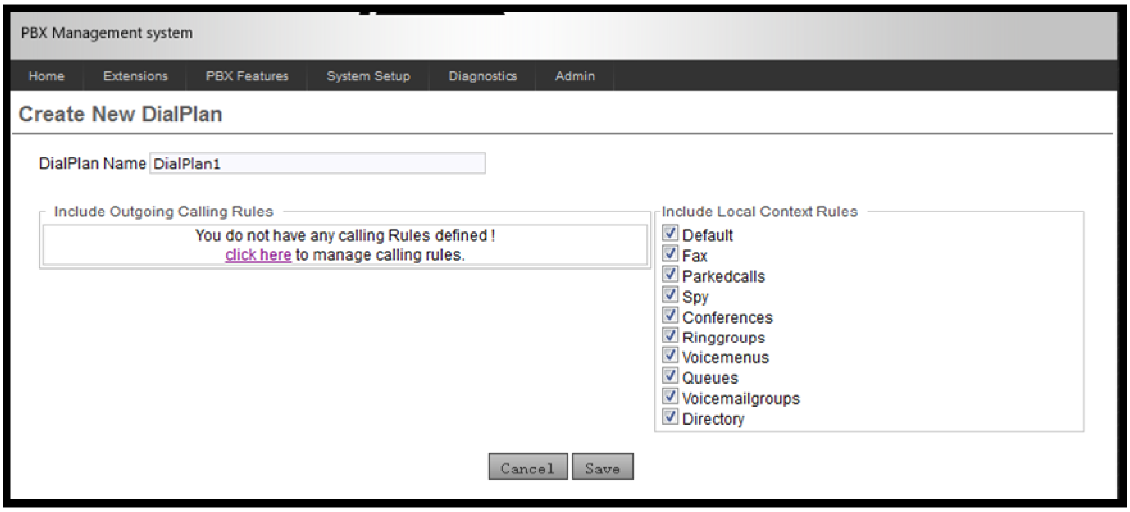


Name	Description	Type	default
Trunk	Choice the trunk for the incoming rule	{analog, server provider, voip}	
Time Interval	Choice the time interval for the incoming rule	Choice	Non timeinterval
Pattern	Pattern of the incoming rule	Dialplan matched	S
Destination	Incoming to destination	{users, voice mail, ring group…}	IVR

1. A trunk support a number of this time intervals, to support a number of Destination
2. Pattern:
All patterns are prefixed by the "_" character. In patterns, some characters have special meanings:
X ... Any Digit from 0-9
Z ... Any Digit from 1-9
N ... Any Digit from 2-9
[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
. Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.
For example, the extension _NXXXXXX would match normal 7 digit dialings, while
_1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.
3.Note:users will most likely need to add a rule with the pattern "s" (without the quotation marks) for each trunk. This signifies 'catch all', meaning all calls with a DID not matching any other rules will match this.
If users have multiple SIP trunks from the same provider, they will want to set this pattern to whatever you specified as Contact Extension.

1.313 Dial Plan

A DialPlan is a set of Calling Rules that can be assigned to one or more users. Please select the Dial Plans option, Click on New DialPlan button, the following table displays the parameters of Dial Plans .

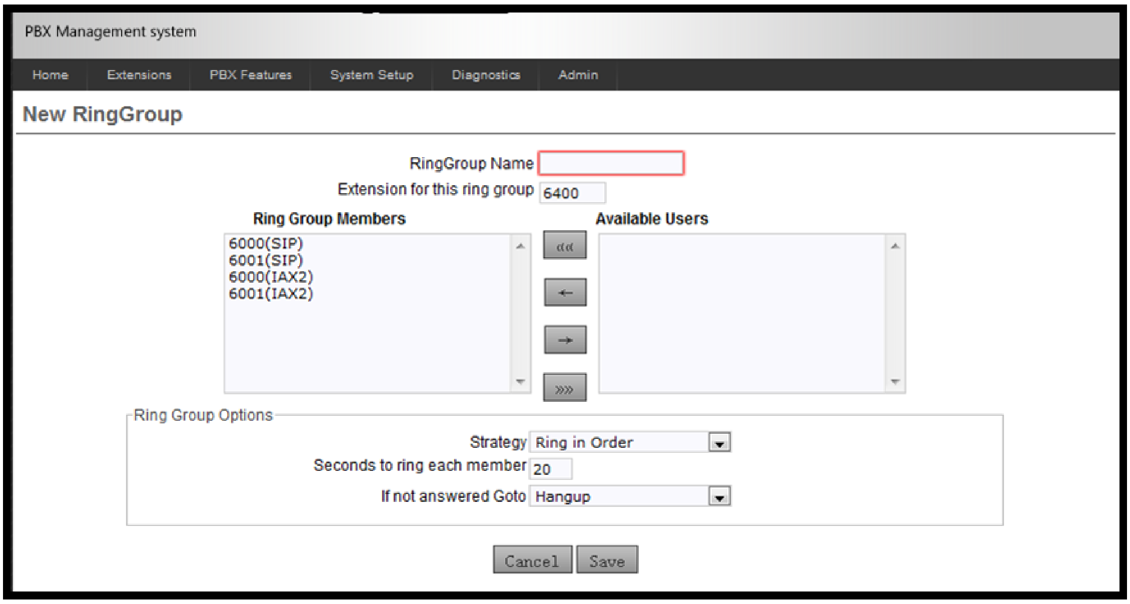


Name	Description	Type	Default
DialPlan Name	The name of DialPlan, which is a unique label to help you identify the dial plan	Textbox	DialPlan1
Include Outgoing Calling Rules	Select outgoing call rule which you use	selected	Not seclect

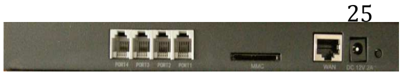
Include Local Contexts Rutes	Local context is used for general using configuration.	check box	Select all

3.314 RingGroups

Define Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Hunt groups.
Please select the Ring Groups option from the vertical menu on the left of the main page, then they can get the following screen:



Name	Description	Type	default
Ring Group Name	Ring group name use in pbx	Str*	
Extension for this ring group	Ring group No.，dial the No. if you want to join ，change boundary value in options	Int	6400
Ring Group Members	The ring group of numbers	{EXT1,EXT2,EXT3,…}	
Available Users	The entire Users	{EXT1,EXT2,EXT3,…}	
Strategy	Ring all simultaneously: Ring in order	{ Ring in Order ,ting all Extensions }	Ring in Order
Seconds to ring each member	Seconds to ring each member	Time	20

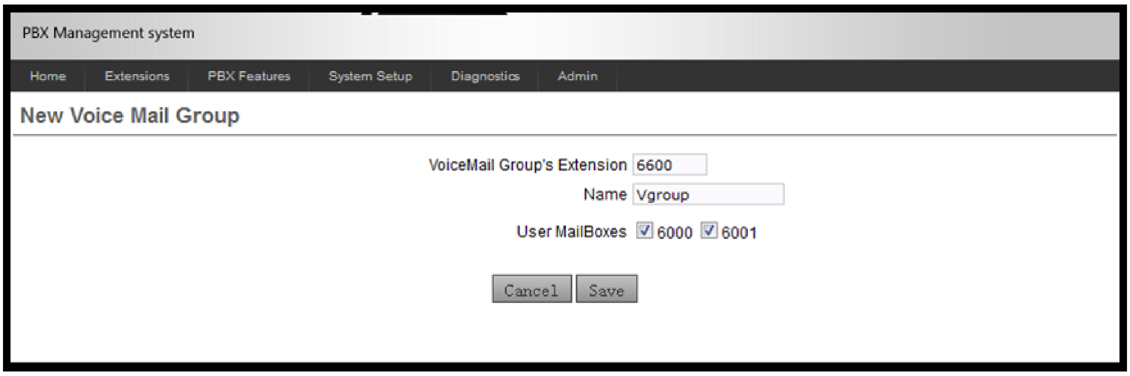


If not answered Goto	If not answered go to, hang up: hang up the calling channel。 Operator：Go to operator 。Extension: a call to user. Voicemail: Go to IVR 。Conference: join a conference. Call queue: Go to a call queue.	{Hang-up, Operator…}	Hang up

1. ring group application: Dial(channel type/${EXTEN}| channel type/${EXTEN}|20|i)
2. ring group up after please a call
3. non-ring if ring group user off hook or non-user registered
4. only one man can connected in coming call

3.315 VoiceMail Groups

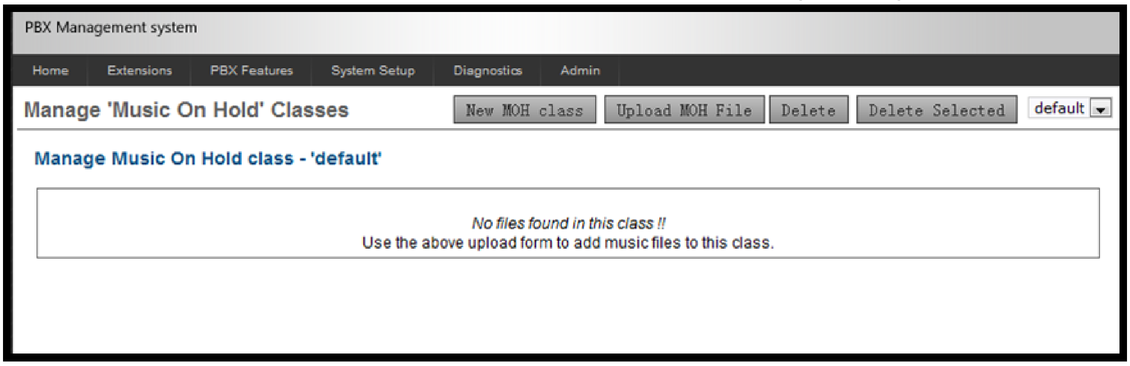
Define Voice Mail Groups to leave a voicemail message for a group of users by dialing a extension.



Name	Description	Type	default
VoiceMail Group's Extension	Default Voicemail Group's Extension	Int	6601
Label	The label of Voicemail Group's Extension	Str*	
User MailBoxes	The entire user Mailboxes	Check boxs	

3.316 Music on Hold

'Music On Hold' need users customize audio tracks for different queues, parked calls etc.



Name	Description	Type	default
Upload an 8 KHz Mono Music file	Support codec: g711a/g711u	Upload	
New music on hold	Add a new music on hold		

2. Music on hold Dir: /persistent/sounds/moh/

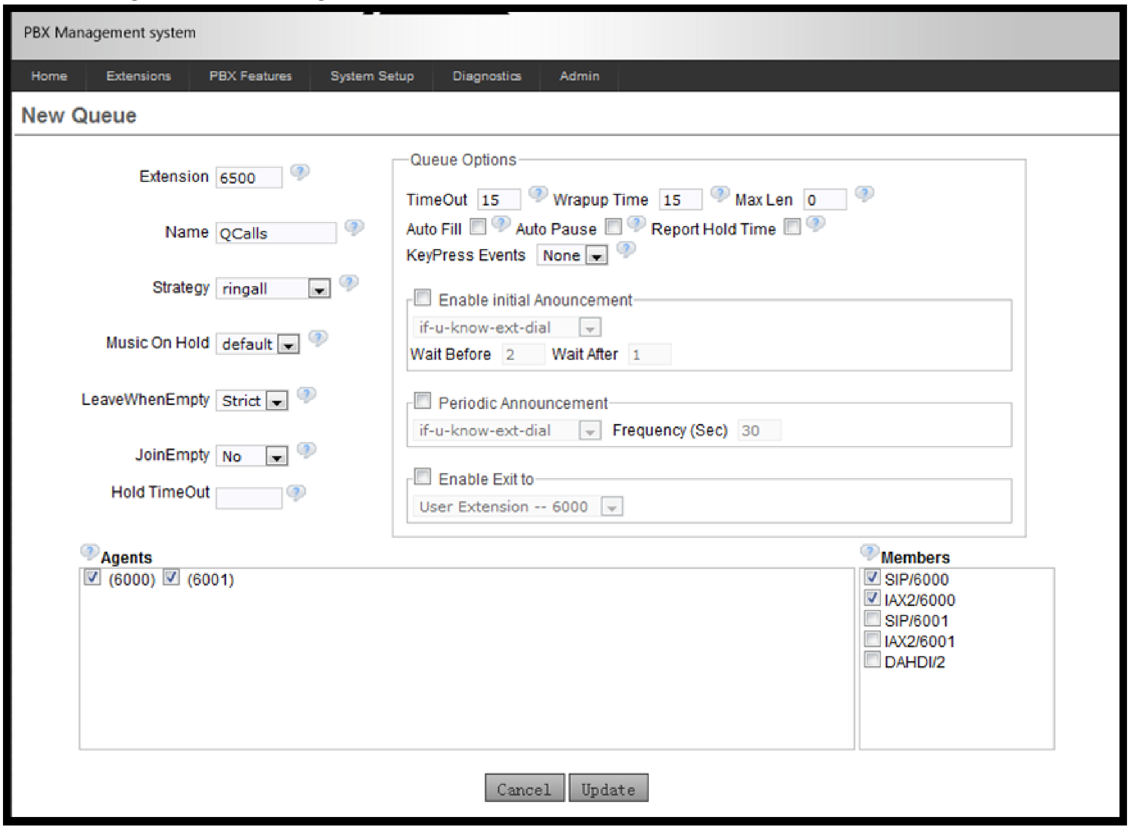
3. Sounds:
   LICENSE-asterisk-moh-freeplay-ulaw
   LICENSE-asterisk-moh-freeplay-ulaw
   fpm-world-mix.ulaw
   fpm-world-mix.alaw
   fpm-sunshine.ulaw
   fpm-sunshine.alaw
   fpm-calm-river.ulaw
   fpm-calm-river.ulaw

4. Music on hold after holding status Status: busy

5. Music on hold non-rtp stream

3.317 Call Queues

Please select the Call Queues option from the vertical menu on the left of the main page, then users can get the following screen



Name	Description	Type	default
Extension	Extension for call queue: may be dialed to reach the call queue	Int	6500
Name	Name for call queue	Str*	
Strategy	Strategy: this option sets the ringing strategy for this queue, the options are 1. Ring all: ring all available agents simultaneously until one answers. 2. RoundRbin: Take turns ringing each available agent. 3. LeastRecent: Ring the agent which was least recently called 4. FewestCalls: Ring the agent with the fewest completed calls 5. Random: Ring a Random agent 6. RRmemory: RoundRobin with Memoryn	{ringall,Roundrobin,le astrecent, F ewest calls,Random,Rrmom ery}	ring all
Music On Hold	Select the 'Music on Hold' Class for this Queue. 'Music on Hold' classes can be managed from the the 'Music On Hold' panel on the left	Choice	default

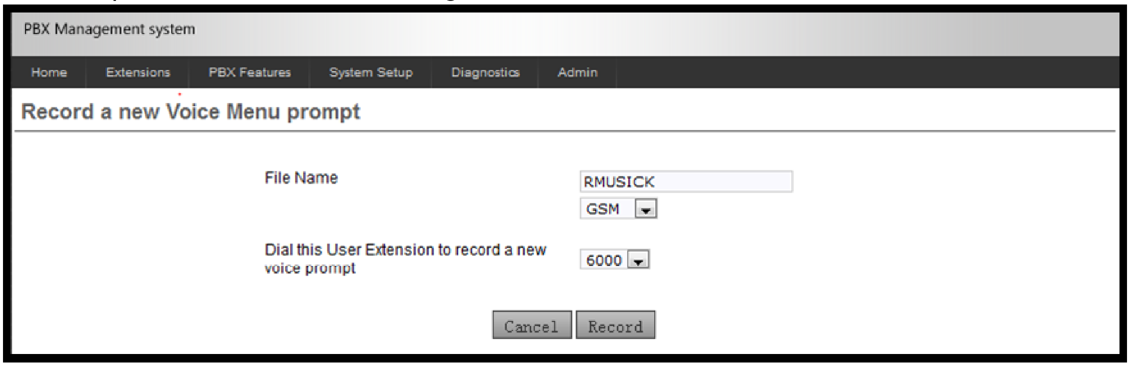
LeaveWhen Empty	This option controls whether callers already on hold are forced out of a queue that has no agents. There are three options. Yes: Callers are forced out of a queue when no agents are logged in. No: Callers will remain in a queue with no agents. Strict: Callers are forced out of a queue with no agents logged in, or if all logged in agents are unavailable. The default option is Strict. After a caller has left the queue, a caller will hear a busy tone and advance to the next calling rule after attempting to enter the queue	{yes,strict , No,}	strict
JoinEmpty	This option controls whether callers can join a call queue that has no agents. There are three options, Yes: Callers can join a call queue with no agents or only unavailable agents No: Callers cannot join a queue with no agents Strict: Callers cannot join a queue with no agents or if all agents are unavailable.	{yes,strict , No,}	no

TimeOut	How many seconds an Agent's phone will ring before the Queue tries to ring the next Agent	Time	15
Wrapup Time	How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 0, which is no delay	Time	0
Max Len	How many calls can be queued at once. This count does not include calls that have been connected with Agents, it only includes calls that have not yet been connected. Default is 0, which is no limit. When the limit has been reached, a caller will hear a busy tone and advance to the next calling rule after attempting to enter the queue	Int	0
Auto full	Defining this option causes the Queue, when multiple calls are in it at the same time, to push them to Agents simultaneously. Thus, instead of completing one call to an Agent at a time, the Queue will complete as many calls simultaneously to the	checkbox	
Auto pause	Enabling this option pauses an agent if they fail to answer a call. This means that the agent is still logged into the queue, but they will not receive calls from the queue. Once paused, an agent can unpause by logging into the queue using the regular agent	checkbox	
Report Hold Time	Enabling this option causes Asterisk to report, to the Agent, the hold time of the caller before the caller is connected to the Agent.	checkbox	
KeyPress Events	If a caller presses a key while waiting in the queue, this setting selects which voice menu should process the key press	choice	
Agent	This selection shows all Users defined as Agents in their User conf. Checking a User here makes them a member of the current Queue	checkbox	

1. Call queue application: Queue(${EXTEN})
2. Change agents status:Login / Login out agents in System Info
3. Hear the music if all agents are busy, until non-conversation busy.

3.318 Voice Menu prompts

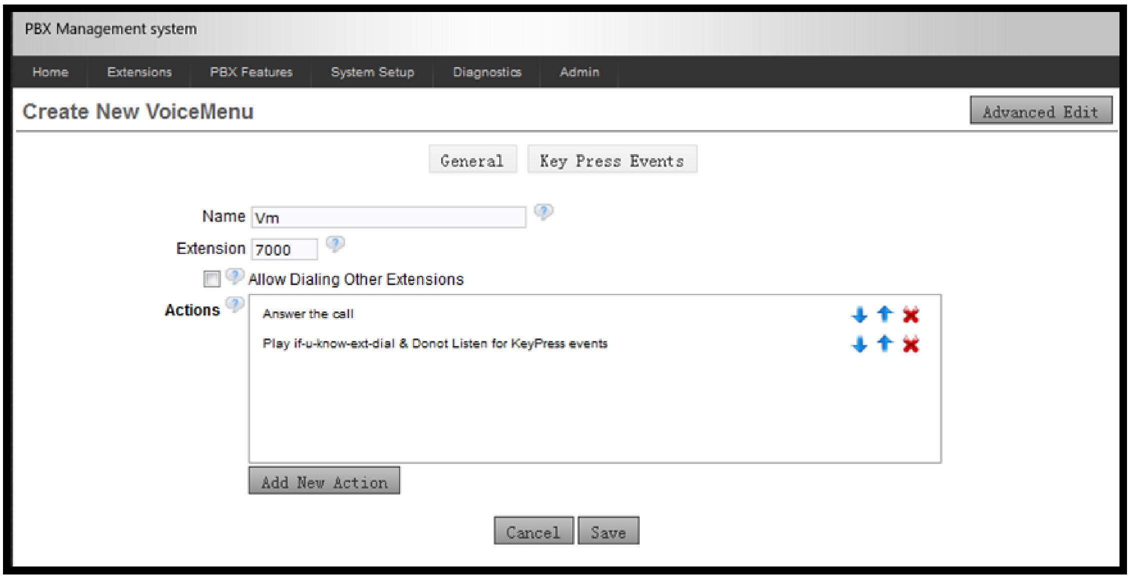
This component is used for recording custom voice menu.



Name	Description	Type	default
Voice menu prompts	File Name	Str*	RMUSIC
	dial this User Extension to record a new voice	Choice	6001
	Voice codes	Choice	

3.319 VoiceMenus

Like most organization, users would like to redirect all of the incoming calls automatically. The voice menu is very handy for these sorts of things. The system should allow callers to make the selection according to the voice menu.



Name	Description	Type	default
Name	A name for the voice menus	Str*	
Extension	If you want this Voicemenu to be accessible by dialing an extension, then enter that extension number	No.	7001
Actions	A sequence of actions performed when a call enters the menu	Dial plan script	
Add new Step	Add additional steps performed during the menu	Dial plan script	
Allow KeyPress Events	Allow key press events will cause the system to listen for DTMF input from the caller and define the actions that occur when a user presses the corresponding digit	checkbox	
Advance edit	Advance edit for the voice menu	Dial plan script	

1. Menus allow for more efficient routing of calls from incoming callers. Also known as IVR (Interactive Voice Response) menus or Digital Receptionist.
2. Step
   a) Answer: Answer a channel if ringing
   b) Authenticate: This application asks the caller to enter a given password in order to continue dialplan execution.
   c) Background: Play an audio file while waiting for digits of an extension to go to. d)Busy Tone: Indicate the Busy condition
   d) Congestion: Indicate the congestion condition to the calling channel. f) Digit Timeout: set digit timeout
   e) DISA Password: Allow someone from outside the telephone switch (PBX) to obtain an internal system dialtone and to place calls from it as if they were placing a call from within the switch.

   f) Response Timeout: set response timeout
   g) Macro: macroname|arg1|arg2 .. Executes a macro using the context 'macro-<macroname>'
   h) Play Sound: Plays back given file k)Ringing: Indicate ringing tone
   i) Set MusicOhHold Class: select a music on hold
   j) SayAlpha: Say each character in the string including letters, numbers and other characters, one by one

k) SayDigits: Say the digits, one by one
l) SayNumber: Say a number (e.g. 'six thousand, five hundred and seventy two')
m) Wait: Pause dialplan execution for a specified number of seconds
n) WaitExten: Wait for the user to enter a new extension for a specified number of seconds r)To Destination: go to destination
o) Set Language: set language (English/Spanish/French)
p) To Directory: go to directory
q) Dial an external Number: Place a call outside the pbx using the selected trunk v) AGI: Executes an AGI compliant application
r) User Event: Send an arbitrary event to the manager interface x) Hangup: Hang up the calling channel

3. Allow keypress events:Must be voice menus have application: Background(file) e.xBackground a music when keypress events
4. Advance edit

Change dialplan for voice menus e.x.

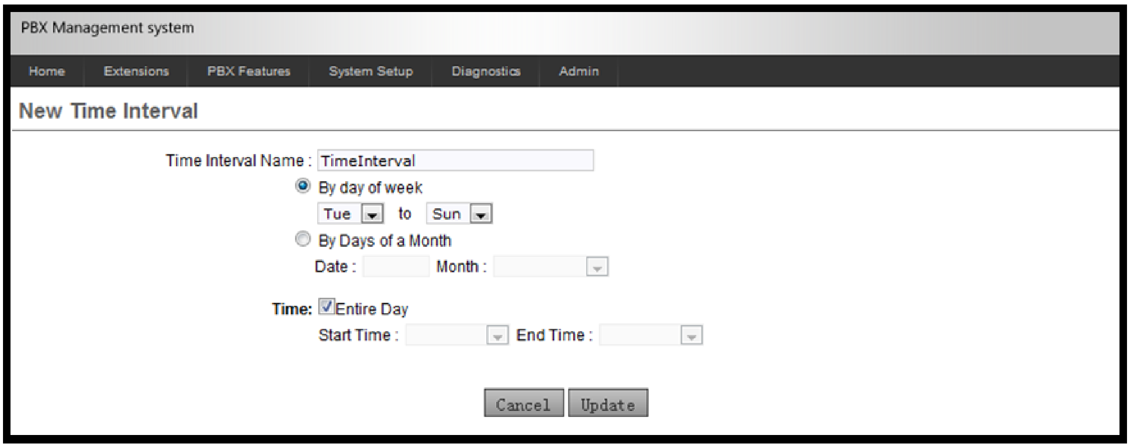
```
include = default
exten = s,1,NoOp(Incoming DID)
exten = s,2,Answer()
exten = s,3,Background(record/GreetingNew)
exten = s,4,Background(record/MakeYourSelection)
exten = s,5,Background(fpm-sunshine)
exten = s,8,Voicemail(6002,u)
exten = 1,1,Goto(voicemenu-custom-2|s|1)
exten = 2,1,Voicemail(6002,u)
exten = 5,1,Goto(voicemenu-custom-3|s|1)
```

Want to control music on hold play time

```
include = default
exten = s,1,NoOp(Incoming DID)
exten = s,2,Answer()
exten = s,3,Background(record/GreetingNew)
exten = s,4,Background(record/MakeYourSelection)
exten = s,5,Set(TIMEOUT(absolute)=8) exten = s,6,Background(fpm-sunshine) exten =
s,7,Set(TIMEOUT(absolute)=60) exten = s,8,Voicemail(6002,u)
exten = 1,1,Goto(voicemenu-custom-2|s|1)
exten = 2,1,Voicemail(6002,u)
exten = 5,1,Goto(voicemenu-custom-3|s|1)
```

3.320 Time Intervals

Time Intervals defines ranges of working time that will be used by call routing features. Please select the Time Intervals option from the vertical menu on the left of the main page,

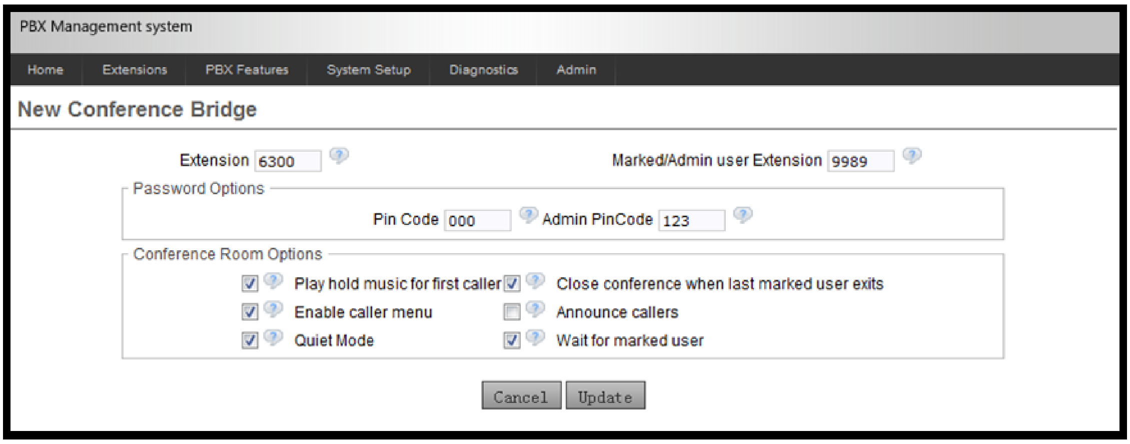


Name	Description	Type	default
Time Interval Name	A name for the time interval	Str*	
By day of week	Choice an available day of week for the time interval	{Mon,Tue, Wed,Thu,Fri ,Sat,Sun }	
By Days of a Month	Choice some available days of month for the time interval	{Dateof January/February/March/April/May/June/july/August/September/October/november/December/all}	
Time	Choice an available time slot for the time interval	{00:00-24:00}	

1. Time intervals using in incoming call
2. Time intervals application rule:
00:00-24:00|mon-sum|1-31|January/February/March/April/May/June/july/August/September/October/
november/December/all
time intervals:
timeinterval_date = *|mon-tue|*|*   Monday to Tuesday of weekly

3.321 Conference rooms

The conferencing function of Asterisk is similar to a Tele-conference call where multiple callers can call in and participate in a two-way conference like in a party room where everyone can talk and listen to one another or just to listen to a Tele-presentation.



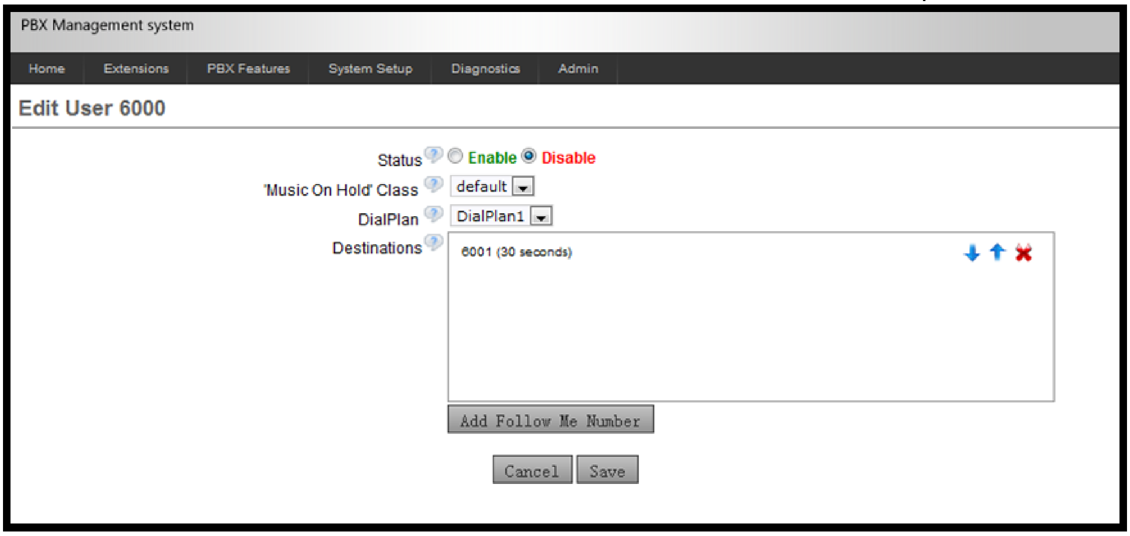
Name	Description	Type	default
Extension	This is the number dialed to reach this Conference	Int	6300
Marked/Admin user Extension	If the conference bridge is to have marked users or admin users, then those users should enter the conference bridge using a separate extension. Admin conference users can lock and unlock the conference and can kick the most recent conference participant. Marked users are special users whose entrance and exit, if the Wait for Marked user or Close conference when last marked user exits can either begin or end the conference altogether	Int	
Pin Code	set an optional pin code, Ex: "1234" that must be entered in order to access the Conference	Str*	
Admin PinCode	Defining this option sets a PIN for Conference Administrators	Str*	
Play music for the first caller	Checking this option causes Asterisk to play Hold Music to the first user in a conference, until another user has joined the same	Check box	unCheck
Close conference for the list caller exit	Close the conference bridge when the last marked user logs out of the conference call	Check box	unCheck
Enable call menu	Checking this option allows a user to access the Conference Bridge menu by pressing the * "Asterisk" key on their dialpad	Check box	unCheck
Announces callers	Checking this option announces, to all Bridge participants, the joining of any other participants	Check box	unCheck
Quiet mode	Do not play enter/leave sounds	Check box	unCheck
Wait for marked user	Prevent conference participants from hearing each other until the marked user has joined	Check box	unCheck

1.Conferencing application:
MeetMe([confno][,[options][,pin]]): Enters the user into a specified MeetMe conference ex.: MeetMe(${EXTEN}|MsIqwxaA)
'1' — disable "you are currently the only person in this conference" message for first member
' a' — set admin mode
'A' — set marked mode

'b ' — run AGI script specified in ${MEETME_AGI_BACKGROUND}
'c' — announce user(s) count on joining a conference
'd' — dynamically add conference
'D' — dynamically add conference, prompting for a PIN
At the pin prompt, if the user does NOT want a pin assigned to the conference, they should hit the # key.
'e' — select an empty conference
'E' — select an empty pinless conference
'F' — Pass DTMF through the conference.
'i' — announce user join/leave with review
'I' --announce user join/leave without review
'M' — enable music on hold when the conference has a single caller
'm' — set monitor only mode (Listen only, no talking)
'p' — allow user to exit the conference by pressing '#'
'P' — always prompt for the pin even if it is specified
'q' — quiet mode (don't play enter/leave sounds)
'r' — Record conference (records as ${MEETME_RECORDINGFILE} using format ${MEETME_RECORDINGFORMAT}).
's' — Present menu (user or admin) when '*' is received ('send' to menu)
't' — set talk only mode. (Talk only, no listening)
'T' — set talker detectio
'v' — video mode
'w' — wait until the marked user enters the conference (plays music on hold until marked user enters if M is used)
All other connected users will hear MusicOnHold until the marked user enters.
'X' — allow user to exit the conference by entering a valid single digit extension of the context specified in ${MEETME_EXIT_CONTEXT} or the current context if that variable is not defined.
'x' — close the conference when last marked user exits

3.322 Follow Me

If A calls B, B does not answer, the call will be transferred to C who is set up in follow me.



Name	Description	Type	default
Status	Enable/Disable FollowMe for this user	Choice	Disable
'Music On Hold' Class	Music On Hold class that the caller would hear while tracking the user	Choice	Default

DialPlan	DialPlan that would be used for dialing the FollowMe numbers. By default this would be the same dialplan as that of the user	Choice	
Destinations	List of extensions/numbers that would be dialed to reach the user during FollowMe	Destinations	
New FollowMe Number	Add a new FollowMe number which could be a 'Local Extension' or an 'Outside Number'. The selected dialplan should have permissions to dial any outside numbers defined	{Dial Local Extension, Dial Outside Number	
Dial Order	This is the order in which the FollowMe destinations are dialed to reach the user	{Ring after Trying previous extension/nu m ber , Ring along with previous extension/nu m ber}	Ring after Trying previous extension/n u mbe
Follow me Option	Playback the unreachable status message if we've run out of steps to reach the or the callee has elected not to be reachable	Check box	Uncheck
	Playback the unreachable status message if we've run out of steps to reach the or the callee has elected not to be reachable	Check box	Uncheck
	Playback the unreachable status message if we've run out of steps to reach the or the callee has elected not to be reachable	Check box	Uncheck

1.General config file : /etc/asterisk/followme.conf

3.4 System Steup

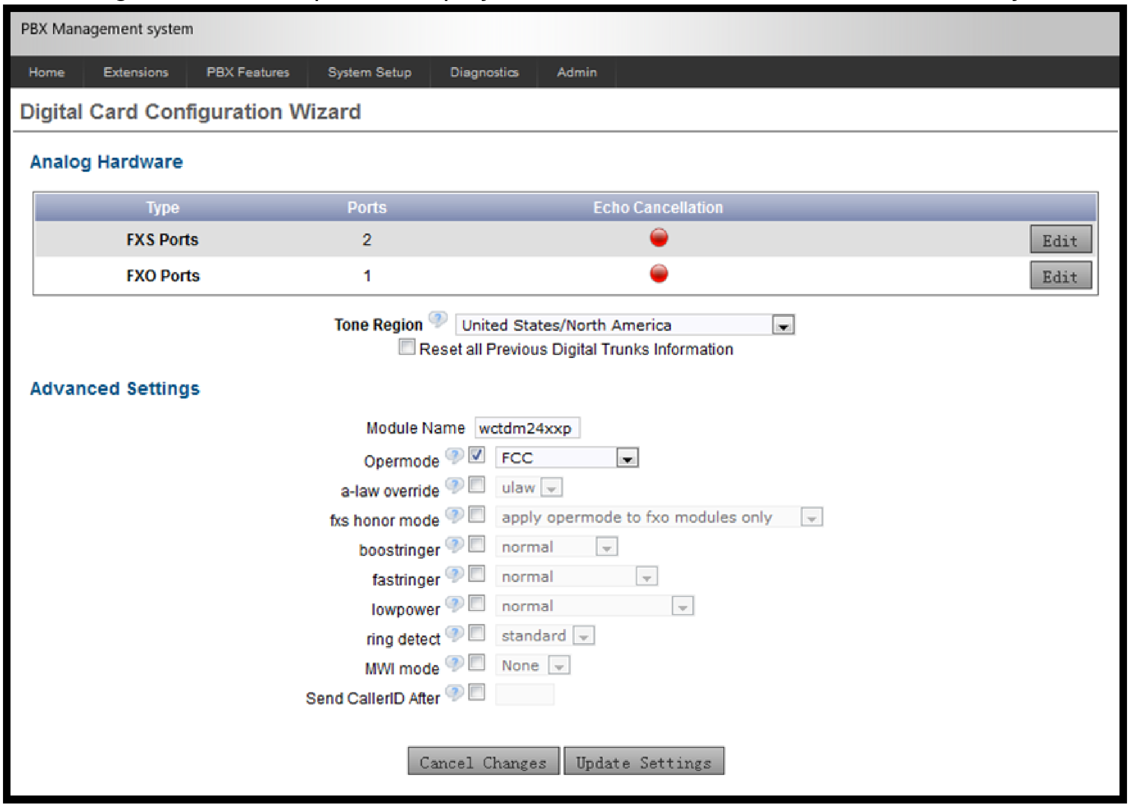
3.411 Configure Hardware

In the configure hardware page, it includes the following components: analog hardware, tone region, advanced settings. Pay attention that some browsers do not display the configure, it is unimportant.

Analog Hardware

When users boot the IP PBX-08, which will detect the FXO and FXS modules automatically,

the analog hardware component displays the modules which are detected correctly.

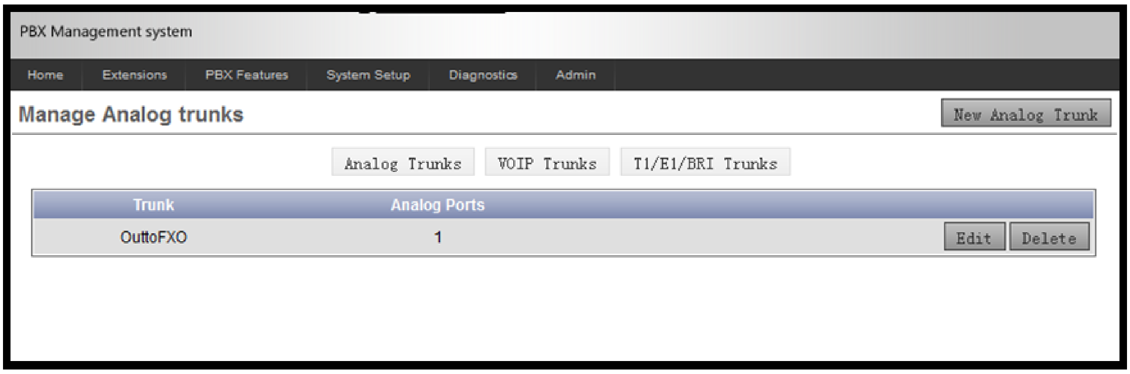


Name	Description	Type	Default
Tone Region	Select the tone region according to your country, if it does not have your country's name in the dropdown list, please ask your service operator which kind of tone region is used in your area	ComboBox	United Status/North America
Module Name	The name of Module	Textbox	wctdm24xxp
Opermode	Specifies On Hook Speed, Ringer Impedance, Ringer Threshold, current Limiting ,TIP/RING voltage adjustment, minimum Operational Look Current and so on. Please choose your country or your nearest neighboring country	ComboBox	USA
a-law override	Specifies the codec to be used for analog	ComboBox	ulaw
fxs honor mode	This option allows the user to determine if they would like opermode characteristics applied to trunk(FXO) modules only, or both trunk (FXO) and station(FXS) modules.	ComboBox	FXO modules
boostringer	This option allows the user to define whether they require normal ringing voltage(40v) or maximum ringing voltage(89v) or analog phones attached to station(FXS) modoules	ComboBox	nomal

fastringer	This option sometimes used in conjunction with the Low Power Option ,allows the user to increase the ringing speed to 25HZ	ComboBox	nomal
lowpower	This option generally used in conjunction with the Fast Ringer Option ,allows the user to set the peak voltage during Fast Ringer Operation to 50V.	ComboBox	nomal
ring detect	This option allows the user to choose from normal ring detection or a full wave detection	ComboBox	standard
MWI mode	This option allows the user to specify the type of Message Waiting indicator detection to be done on trunk(FXO) interfaces	ComboBox	none

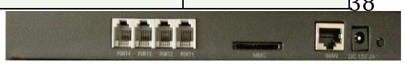
3.412 Configure trunks

To receive calls from PSTN and make calls to the outside world, users have to use trunks. Please select the Trunks option from the vertical menu on the left of the main page.



Analog trunk is associated with FXO port, and it will call outside by PSTN line. Click on New Analog Trunk , then users can see the parameters which are in the following table in the web.

Name	Description	Type	Default
Channels	Display the FXO or FXO/FXS modules	selected	no select
Trunk Name	The name you want to set for the trunk	Textbox	null
Busy Detection	Busy detection is used to detect far end hang up or for detecting busy signal.	Boolean	Yes
busycount	If Busy Detection is enabled,it is also possible to specify how many busy tones to wait for before hanging up.	Int	3
Ring Timeout	Thrunk(FXO) devices must have a timeout to determine if there was a hangup before the line was answered.	Int	8000



answeronpolarityswitch	If this option is enabled, the reception of a polarity reversal will mark when a outgoing call is answered by the remote party.	Boolean	no
hanguponpolarityswitch	In some countries ,a polarity reversal is used to single the disconnect of a phone line	Boolean	no
Use CallerID	Enabling this option enabled CallerId detection.	Boolean	yes
Caller ID Start	This option allows one to define the start of a CallerID Signal.	ComboBox	Ring
CallerID	This option allows the lines to report the Caller ID string as received from the telco, or as a fixed value by using the custom option.	select box	As Received
Pulse Dial	If this option is enabled ,pulse mode dialing instead of DTMF,wil be enable.	Boolean	No
CID Signalling	This option defines the type of caller ID signaling to use :bell,v23,v23_jp,or dtmf.	ComboBox	Bell-USA
Flash Timing	Flash Time defines the time ,in millseconds,that is generated for a flash operation.	Textbox	750
Receive Flash Timing	Flash Time defines the time,in milliseconds, that is generated for a flash operation.	Textbox	1250

1.Trunk name: unique label to help users identify the trunk when listed in outgoing calling rules and incoming calling rules.

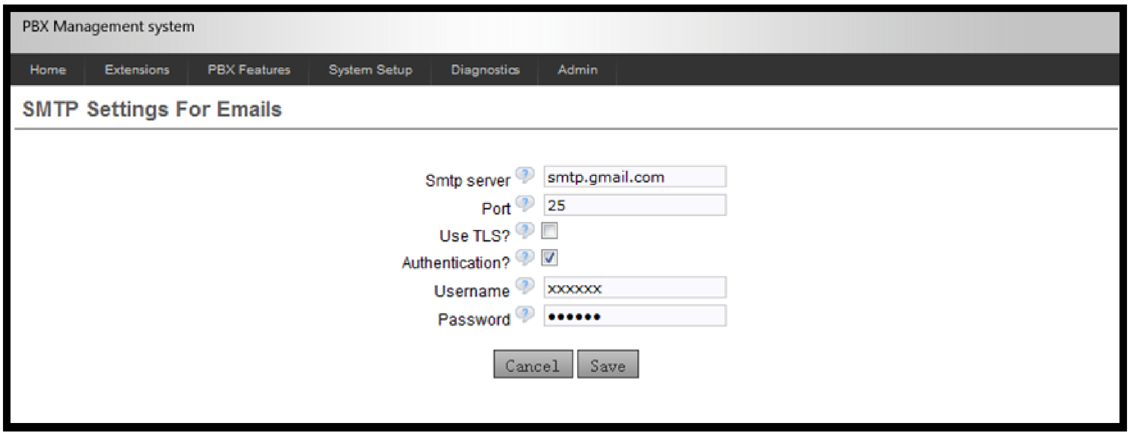
A VoIP service provider (VSP) that users have signed up with is also a trunk. Via the VoIP trunk users can dial via the VoIP service to reduce their cost when making international calls. Users can set up the VoIP trunk to make calls to the PSTN or other VoIP network. Users also can use the VoIP trunk to link headquarter and branch offices for free internal calls.
Click on New SIP/IAX Trunk, the following table is the parameter of VoIP trunk:

Name	Description	Type	Default
Type	You can select SIP or IAX type to meet your need.	ComboBox	SIP
Provider Name	A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc.	Textbox	Null
Hostname	The IP Address of the server which you want to connect	Textbox	Null
Username	the username that your service provider configured	Textbox	Null
Fromdomain	The domain of the server which you want to connect	Textbox	Null

Password	the password that your service provider configured for the user.	Textbox	Null
Contact Ext.		Textbox	s
Insecure Type	The insecure type of the trunk transferring data.	ComboBox	very

1.Notice Provider Name must be unique label , especially do not the same with Username .
2.Insecure Type: insecure=very ; To allow registered hosts to call without re-authenticating insecure=port ; Allow matching of peer by IP address without matching port number. insecure=invite; removes the requirement for authentication of incoming INVITE messages.

3.413 SMTP Setting



Name	Description	Type	default
STMP server	The IP address or hostname of an SMTP server that your box may connect to, without authentication, in order to send e-mail notifications of your voicemails; i.e.	Str*	
Port	The port number on which the SMTP server is running; generally port 25	Str*	
Use TLS?	Use TLS(Transport Layer Security) when communicating with the SMTP server?	Check box	unCheck
Authentication?	Does the SMTP Server requite authentication?	Check box	unCheck
Username	The username of a valid account on the STMP	Str*	
Password	The password of a valid account on the STMP	Str*	

1. Config file: /etc/ssmtp/ssmtp.conf
2. Note: Firmware after that starts support Gmail

3.414 Voicemail Setting

When users call someone who does not answer the call, users can leave a voice message for the called party if the called party supports voice mail.

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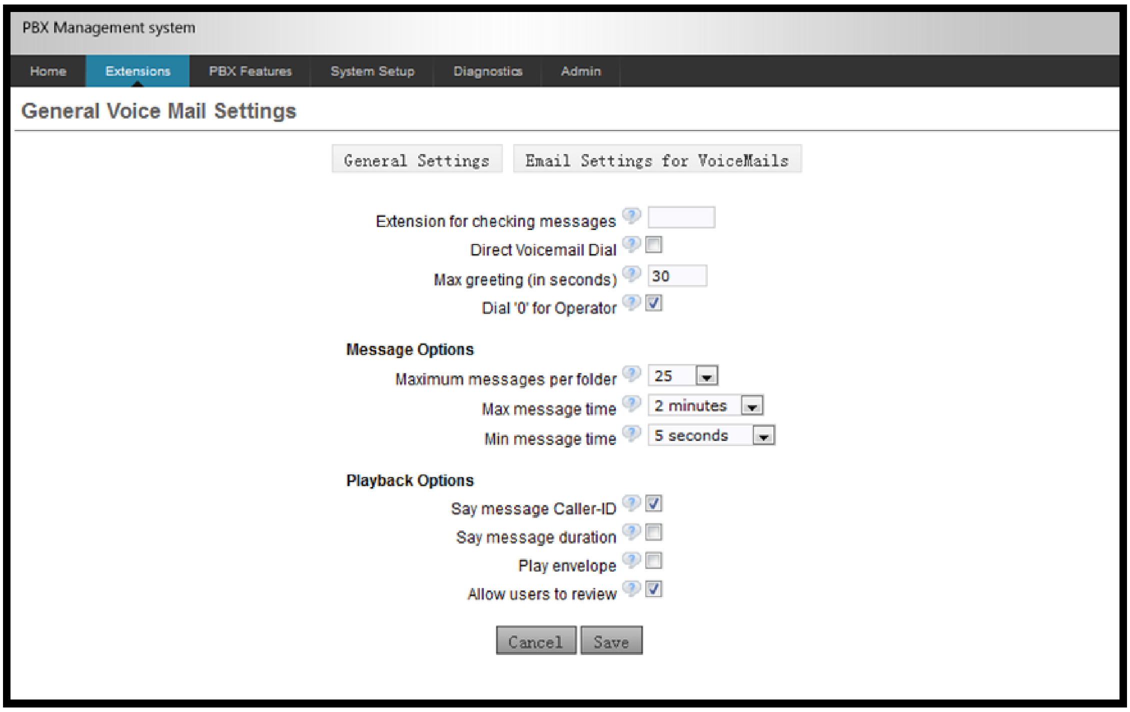


Name	Description	Type	Default
Extension for checking messages	defines the extension that Users call in order to access their voicemail accounts	NO.	6750
Direct Voicemail Dial	Check this to enable direct voicemail dial. For instance, if John's extension is 6001, you would be able to directly dial into John's voicemailbox by dialing #6001 to leave him a message	Check box	unCheck

Max greeting (in seconds)	Set the maximum number of seconds for a User's voicemail greeting	No.	30
Dial '0' for Operator	Enable Callers to exit the voicemail application and connect to an operator extension. The operator extension must be defined from the 'Options' panel	Check box	Check
Maximum messages per folder	This select box sets the maximum number of messages that a user may have in any of their folders	{10,25,100, 200,500,1000}	25
Max message time	This select box sets the maximum duration of a voicemail message in seconds. Message recording will not occur for times greater than this amount	{1 minute ,2 minutes,5 minutes,15 minutes,30 minutes,umlimited}	2 minutes
Min message time	This select box sets the minimum duration of a voicemail message in seconds. Messages below this threshold will be automatically deleted.	{no minimum,1 seconds,2 seconds,3 seconds,4 seconds,5 seconds}	1 seconds
Say message Caller-ID	If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.	Check box	Check
Say message duration	If this option is set, the duration of the message in mintues will be played back before the voicemail message begins playing	Check box	unCheck
Play envelope	Turn on/off playing introductions about each message when accessing them from the voicemail	Check box	unCheck
Allow users to review	Checking this option allows the caller to review their message before it is submitted as a new voicemail message	Check box	Check

1. Voice mail application: ,Voicemail(${ARG},u)
2. Automatically generated configuration file (/etc/asterisk/voicemail.conf)

mailbox_number => password, name, email
mailbox_number : the number you use in extension.conf for VoiceMail() command and to register a user in sip.conf or iax.conf
password : the pass used to register a user in sip.conf or iax.conf
name : the name which to be associated with the mailbox
email : where a notification for the voicemail will come

3. IPPBX Max messages data: 150M

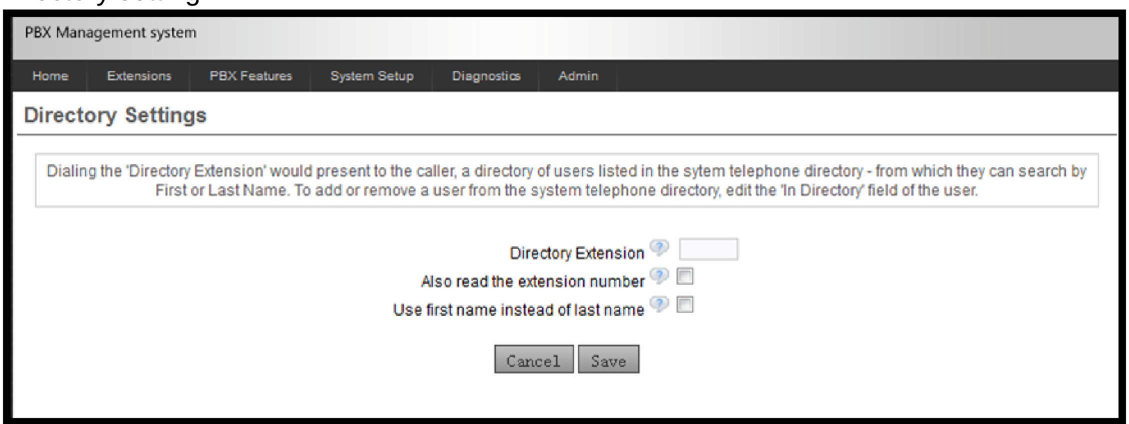
a) Email Settings for Voice mails

Name	Description	Type	default
Send messages by e-mail only	If this option is set, then voicemails will not be checkable using a Phone. Messages will be sent via e-mail, only. Note: You need to have an smtp	Check box	unCheck

Attach recordings to e-mail	This option defines whether or not voicemails are sent to the Users' e-mail addresses as attachments. Note: You need to have an smtp server configured for this functionality	Check box	Check
Template for	From	Str*	
Voicemail Emails			ourcompany.null
	Subject	New voicemail from ${VM_CALLERID} for ${VM_MAILBOX}	
	Template Variables: \t : TAB ${VM_NAME} : Recipient's firstname and lastname ${VM_DUR} : The duration of the voicemail message ${VM_MAILBOX} : The recipient's extension ${VM_CALLERID} : The caller id of the person who left the message ${VM_MSGNUM} : The message number in your mailbox ${VM_DATE} : The date and time the message	Hello ${VM_NAME}, you received a message lasting ${VM_DUR} at ${VM_DATE} from, (${VM_CALLERID}). This is message ${VM_MSGNUM} in your voicemail Inbox.	

3.415 Directory Setting

Dialing the 'Directory Extension' would present to the caller, a directory of users listed in the system telephone directory - from which they can search by First or Last Name. To add or remove a user from the system telephone directory, edit the 'In Directory' field of the user. Preferences for 'Dialing by Name Directory'.
Directory setting:

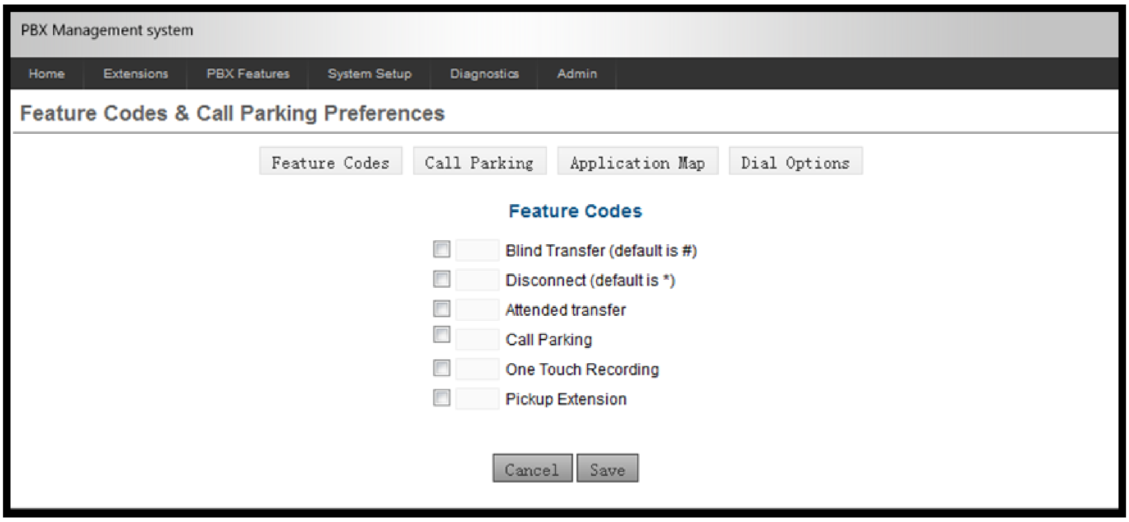


Name	Description	Type	default
Directory Extension	Extension to dial for accessing the Name Directory	Int	
Also read the extension number	In addition to the name, also read the extension number to the caller before presenting dialing options	Check box	Uncheck
Use first name instead of last name	Allow the caller to enter the first name of a user in the directory instead of using the last name	Check box	Uncheck

1. Directory application: Directory(default|default|ef)

3.416 Call Feature

Feature Codes and Call parking preferences
Features Codes



Name	Description	Type	default
Features Codes	Blind Transfer (default is #)	Check box&&Int	#
	Disconnect (default is *)	Check box&&Int	*
	Attended transfer	Check box&&Int	
	Call Parking (Packing a call)	Check box&&Int	

Call Parking Preferences

Name	Description	Type	default
Call Parking Preferences	Extension to Dial to Park a call	Int	700
	What extensions to park calls on	Int	701-720
	Number of seconds a call can be parked for	Time	

Application Map

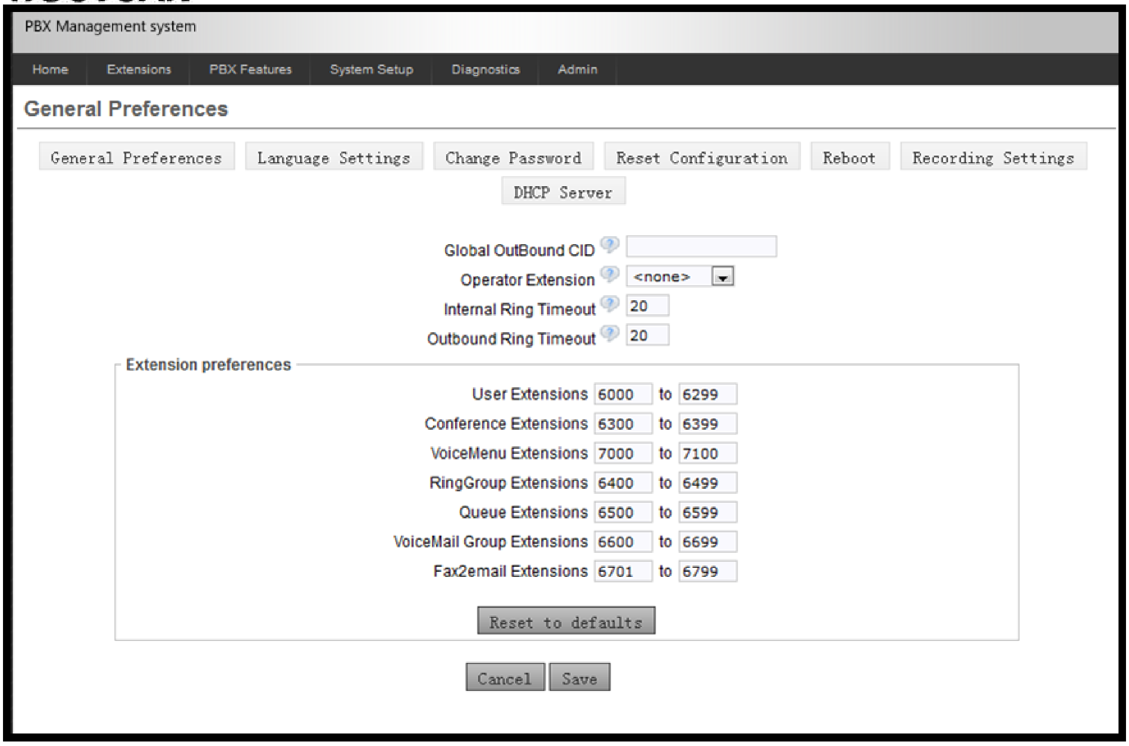
Name	Description	Type	default
Application Map	Add an application for PBX		

Dial Options

Dial Options	(t-Option) Allow the called party to transfer the calling party by sending the DTMF sequence defined on the Feature Codes page	Check box	Uncheck
	(T-Option) Allow the calling party to transfer the called party by sending the DTMF sequence defined on the Feature Codes	Check box	Uncheck
	(h-Option) Allow the called party to hang up by sending the DTMF sequence defined on the Feature Codes	Check box	Uncheck
	(H-Option) Allow the calling party to hang up by sending the	Check box	Uncheck
	(k-Option) Allow the called party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes	Check box	Uncheck
	(K-Option) Allow the calling party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes	Check box	Uncheck

3.417 Options

This component is used for administrator to manage the system, it includes the following modules: General Preferences



Name	Description	Type	default
Global OutBound CID	This is default global CallerID that is used for all outgoing calls when no other CallerID is defined that has a higher priority 。When making outgoing calls the following rules are used to determine which CallerID will be used, if they exist: The first CallerID used is a CallerID set for the user making the call defined in the 'Users' tab. The second CallerID is the one that is set in the 'VoIP Trunks' configuration, if applicable The last CallerID used for outgoing calls is the Global CID defined in the 'Options' tab.	Int	
Operator Extension	The Operator Extension is the extension which will be dialed when a caller presses '0' to exit Voicemail. It is also available as a Voice Menu option	Chioce	
Ring Timeout	Number of seconds to ring a device before sending to the user's Voicemail Box	Time	20
Call Record Dir	Call Record Dir	Str*	/tmp
Call Record Format	Call Record Format	Choice	FXO/FXS

Extension preferences	User Extensions	Int	6001-6299
	Conference Extensions	Int	6300-6399
	VoiceMenu Extensions	Int	7001-7100
	RingGroup Extensions	Int	6400-6499
	Queue Extensions	Int	6500-6599
	VoiceMail Group Extensions	Int	6600-6699
	Resert to default		

Languages

Name	Description	Type	default
Languages	The Language setting allows the user to specify the default prompts language for phone to phone, inbound, and outbound calls. If a soundpack selection is made but not already installed, then the pack will be downloaded from Digium	Chioce	English

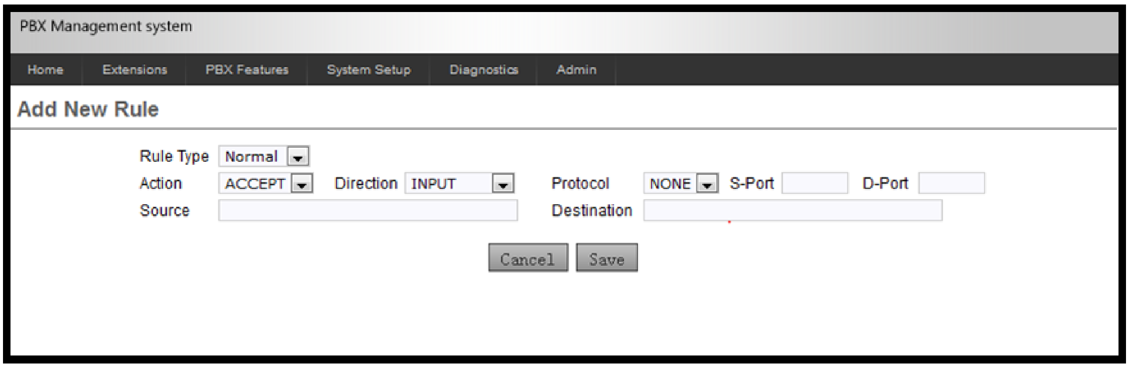
Change Password

Name	Description	Type	default
Change Password	Enter New Password	Str*	
	Retype New Password	Str*	

Factory reset

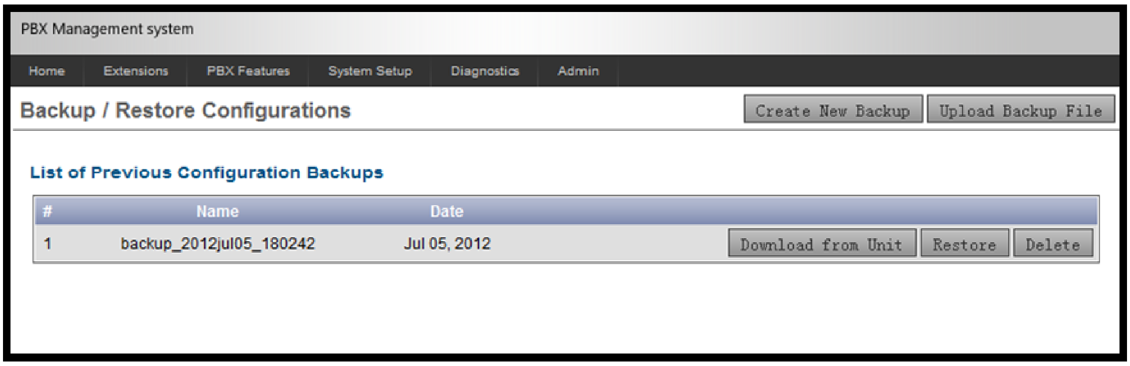
Name	Description
Factory reset	Reset to defaults include network settings
	Reset to defaults but keep network settings

3.418 IP table Firewall



3.419 Backup

Backup and Restore are two of the mandatory functions of any application. IP PBX-02\04\08 is no exception. Customers can backup all the files under the /etc/asterisk/ directory and restore them.

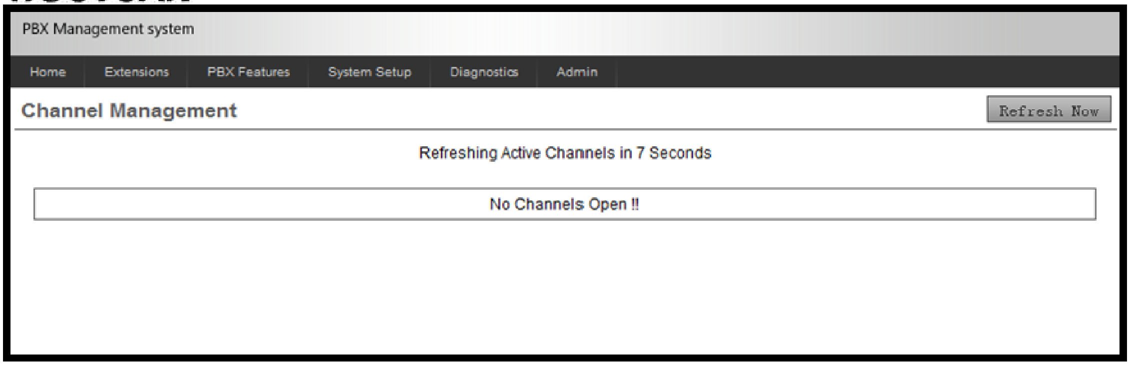


Name	Description	Type	default
Backup	Create new backup		
	Download from Unit		
	Restore Previous config		

3.5 Diagnostics

3.51 Active Channels

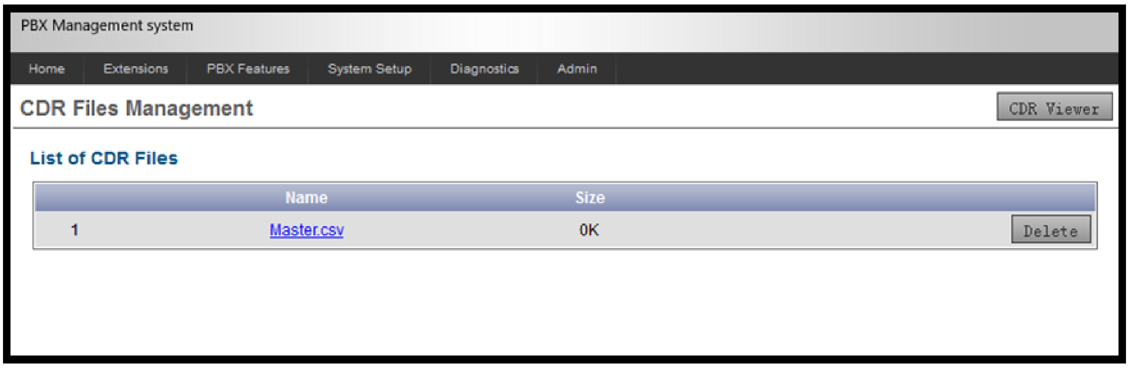
The channels which are in communication status will be displayed in this component.



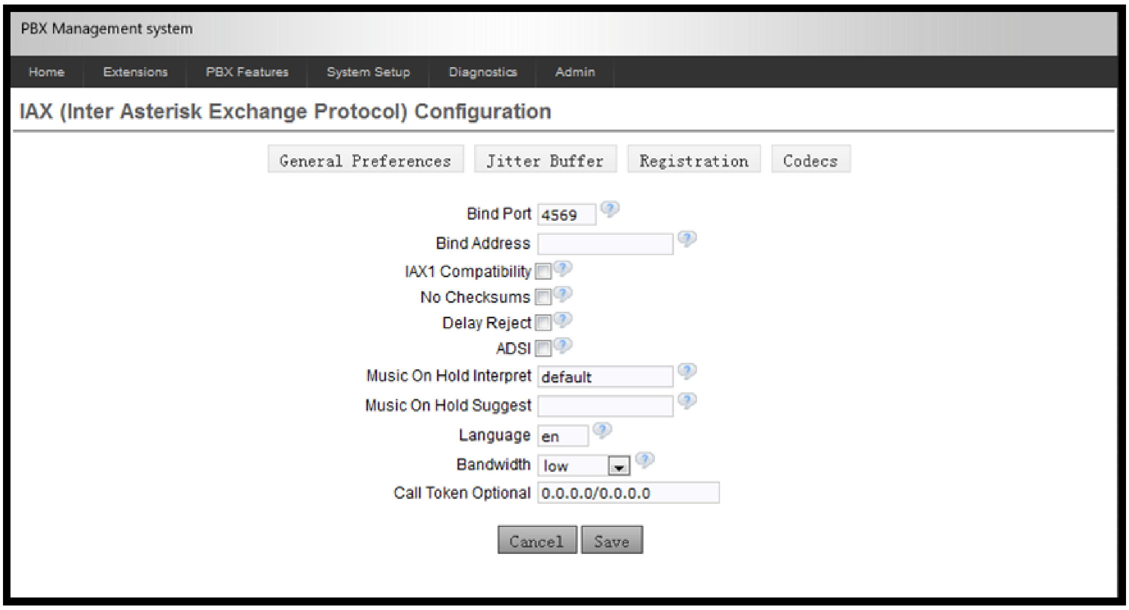
Refresh Now	Description
Status	Upload message for asterisk channels Hangup：hang-up channel Transfer：transfer channel

3.6 Admin

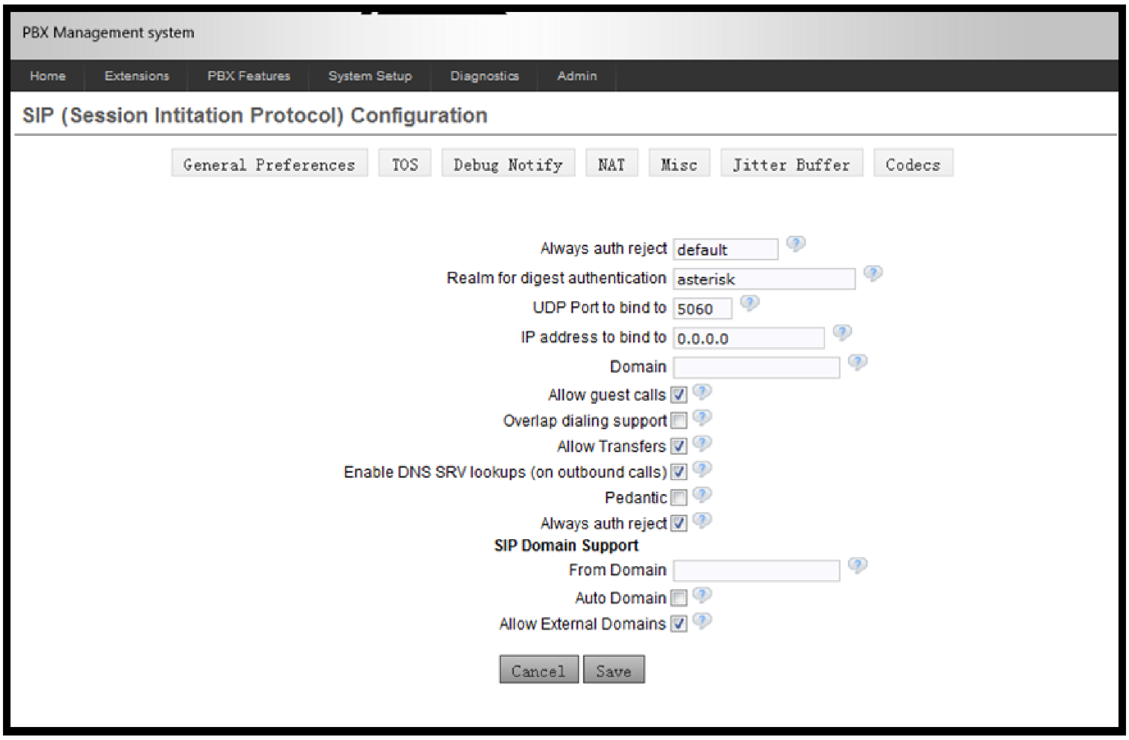
3.61 CDR Viewer



3.62 IAX Setting



3.63 SIP Setting



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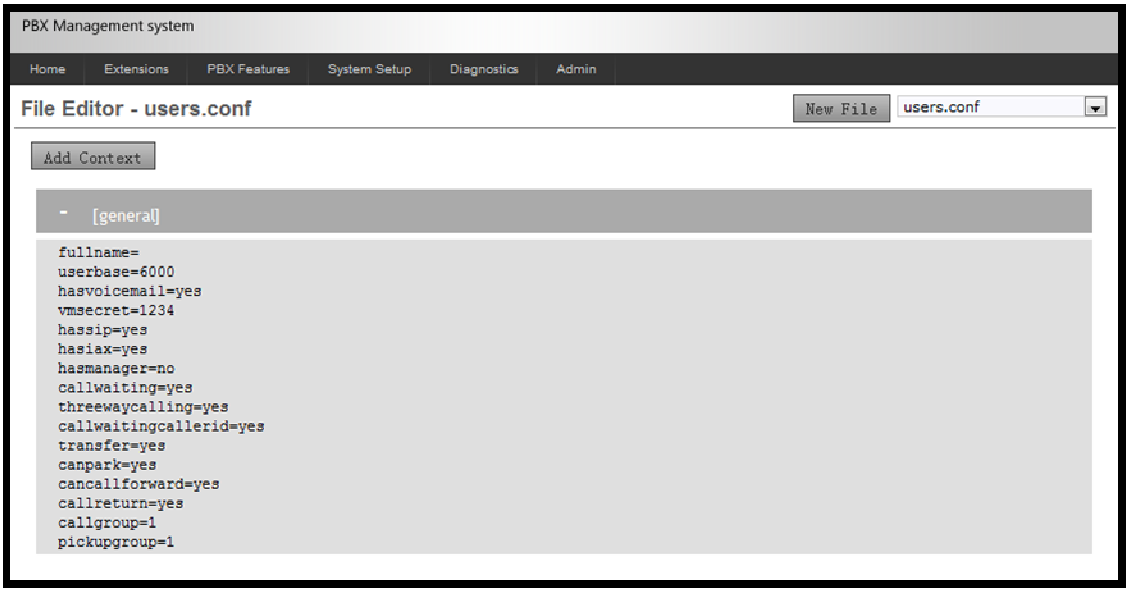
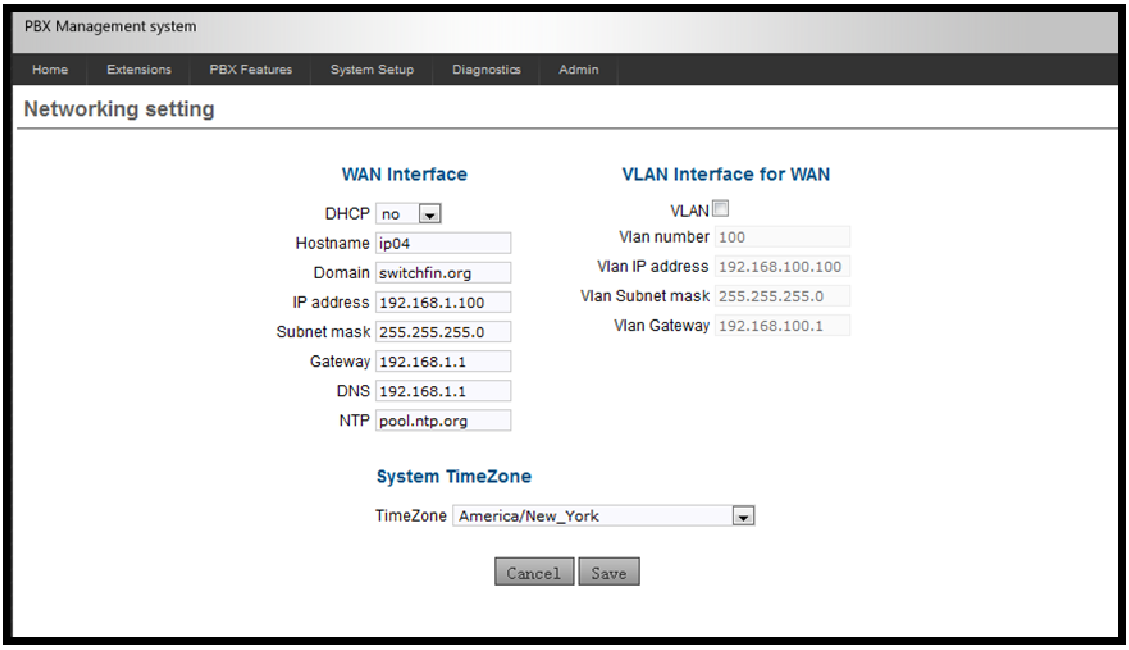
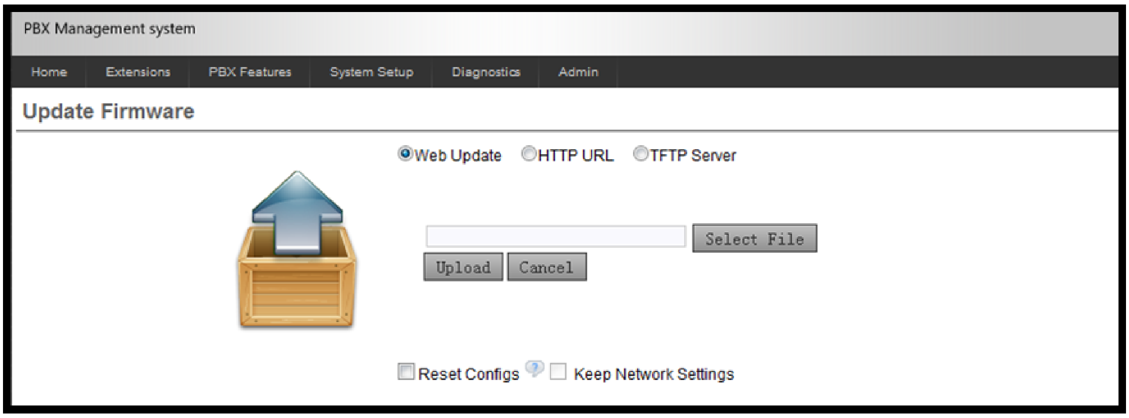
3.64 File Edit



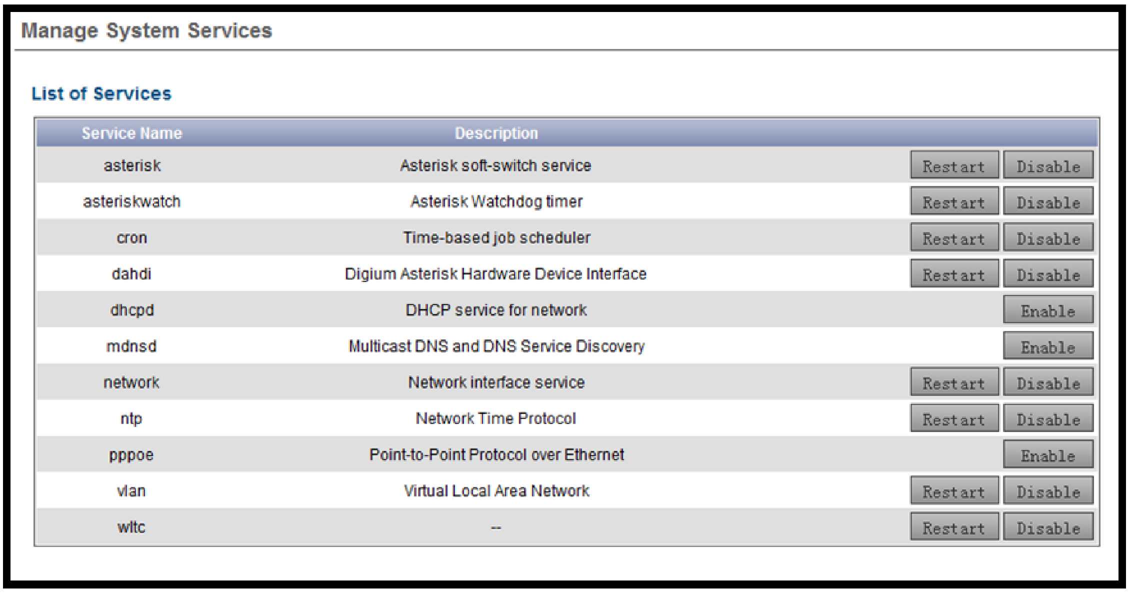
3.65 Network Setting



3.66 Firmware



3.67 Server Message

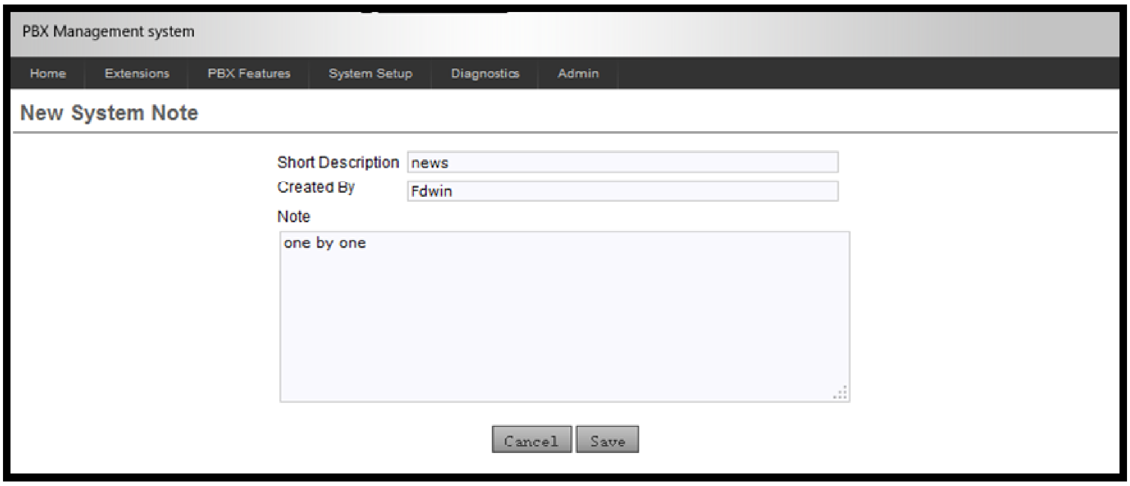


Manage System Services

List of Services

Service Name	Description		
asterisk	Asterisk soft-switch service	Restart	Disable
asteriskwatch	Asterisk Watchdog timer	Restart	Disable
cron	Time-based job scheduler	Restart	Disable
dahdi	Digium Asterisk Hardware Device Interface	Restart	Disable
dhcpd	DHCP service for network		Enable
mdnsd	Multicast DNS and DNS Service Discovery		Enable
network	Network interface service	Restart	Disable
ntp	Network Time Protocol	Restart	Disable
pppoe	Point-to-Point Protocol over Ethernet		Enable
vlan	Virtual Local Area Network	Restart	Disable
witc	--	Restart	Disable

3.68 Server Notes



Case of IP PBX-02\04\08

Figure: Network Topology

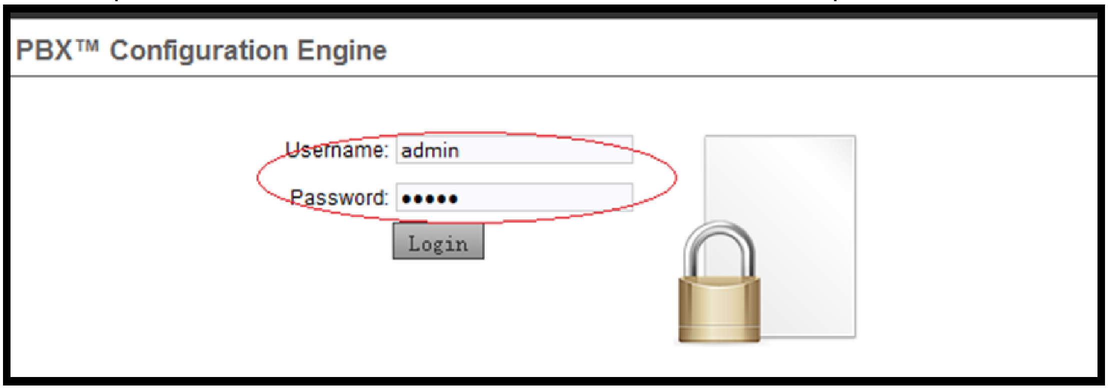
In the network topology above: user 6020,6001,6002,6008 will be registered to IP PBX-02\04\08, After configuration, it will realize the following function:

1) The internal user 6002 and user 6001 can call each other directly.
2) 6001, 6002, 6008 can communicate with outside through IP PBX-02\04\08 by FXO/FXS.
3) User 6001 and 6030 can call each other through VoIP trunk, although they are registered to different IP PBX.
4) User 6020 and 6001 can call each other directly, although they are not in the same network segment.
5) Voicemail
6) IVR
7) Conference
8) Ring Groups
9) Agents
10) Follow me
11) Call pickup

How to Make Internal Calls through IP PBX-02\04\08

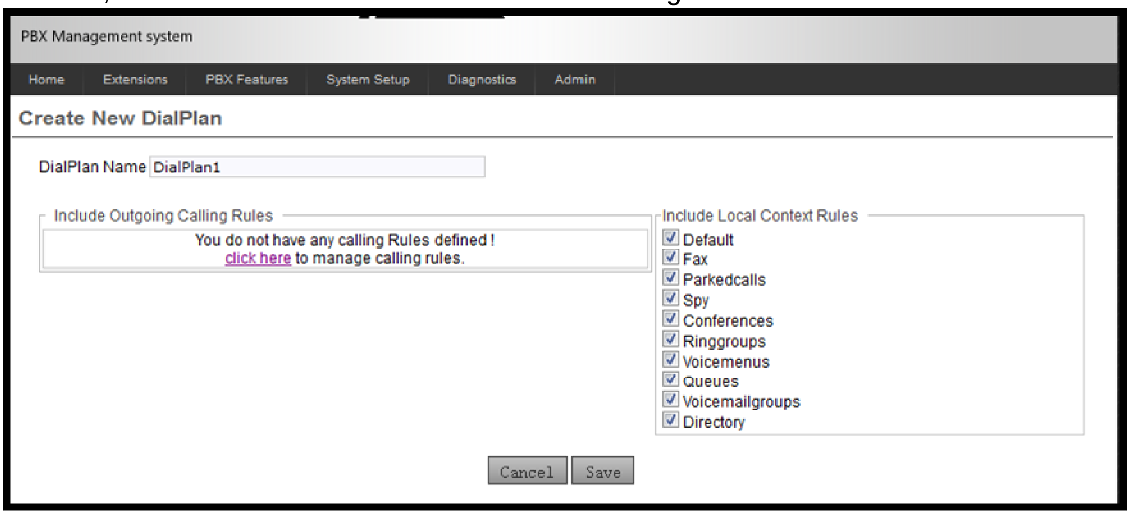
Access to the Web Page of IP PBX-02\04\08 by Browser

After connecting IP PBX-02\04\08 to LAN, please open your browser of PC with OS and input the IP Address of IP PBX-02\04\08 (the default IP address is 192.168.1.167)
Please input the default Username: admin; Password: admin in the presented screen above.



Add up Users from Web Page of IP PBX-02\04\08

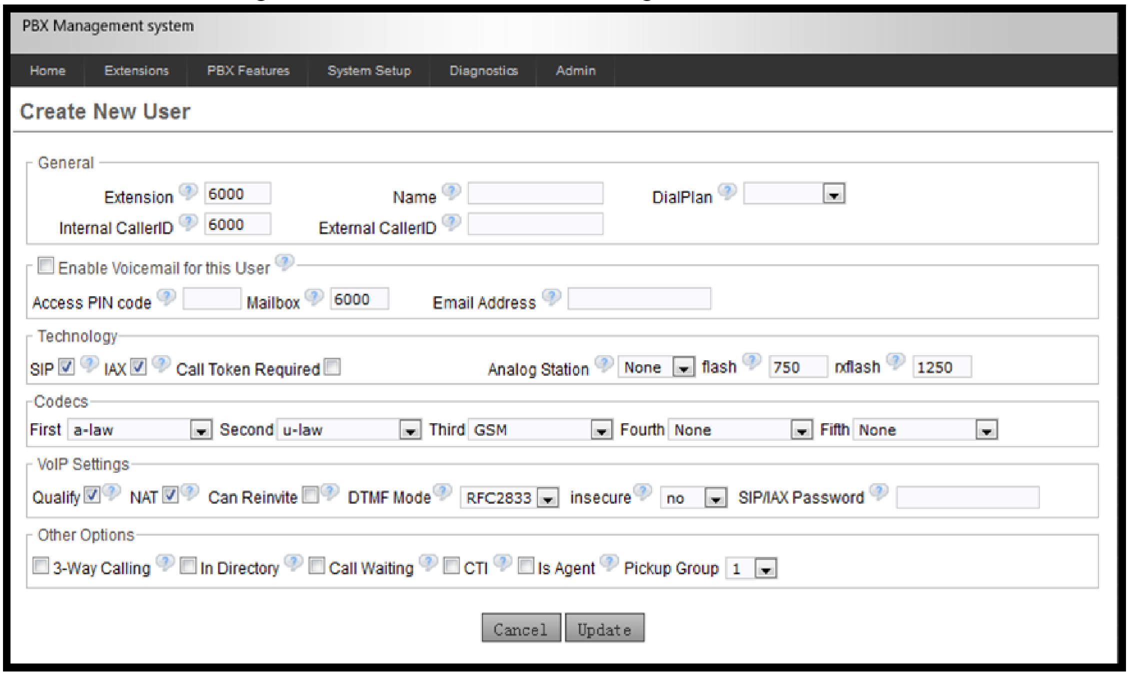
First: Add up a DialPlan
Before users add up user, they have to add up a DialPlan, please click on Dial Plans New DialPlan, the writer creates a DialPlan like the following:



After configuring, please click on Save button, and click on Apply Changes button in up right corner of the main page

Next: Add up SIP user 6000

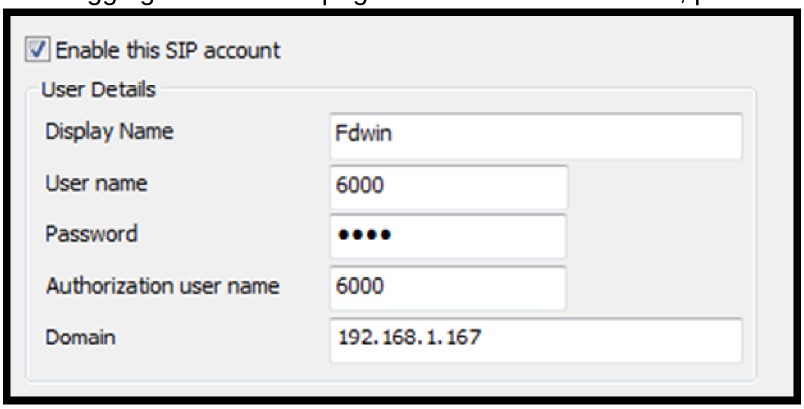
After logging into the web page of IP PBX-02\04\08, please click on Users Create New User, the writer configure user 6000 like the following :



At last, please click on Update button, and click on Apply Changes button in up right corner of the main page.

Register a SIP user 6000 in IP PHONE

After logging into the web page of IP Phone IP PHONE, please select VOIP option,



After configuring, please click on the APPLY button. Users can see the “Register status” is Registered, if user do not register successfully, please pay attention to the Password in the red ellipse frame , which must be the same with the SIP/IAX Password of the user 6001 in IP PBX-02\04\08.

Now users can call each other directly between user 6001, 6002 and 6008.

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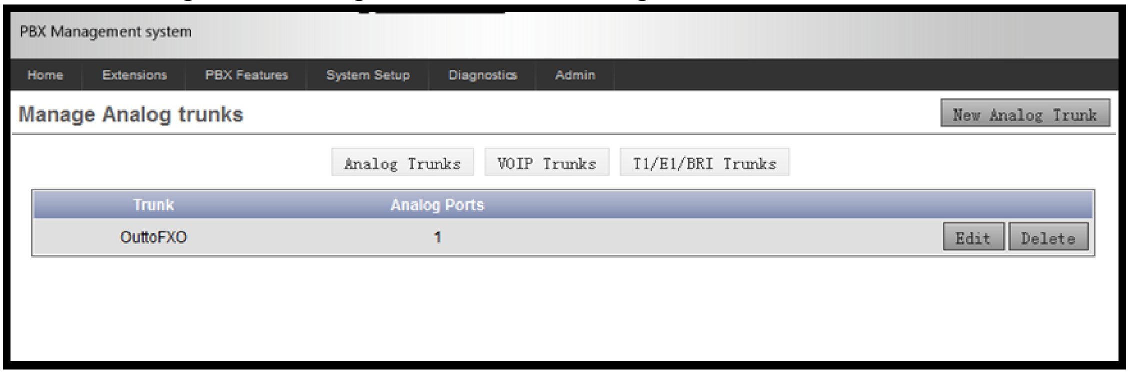
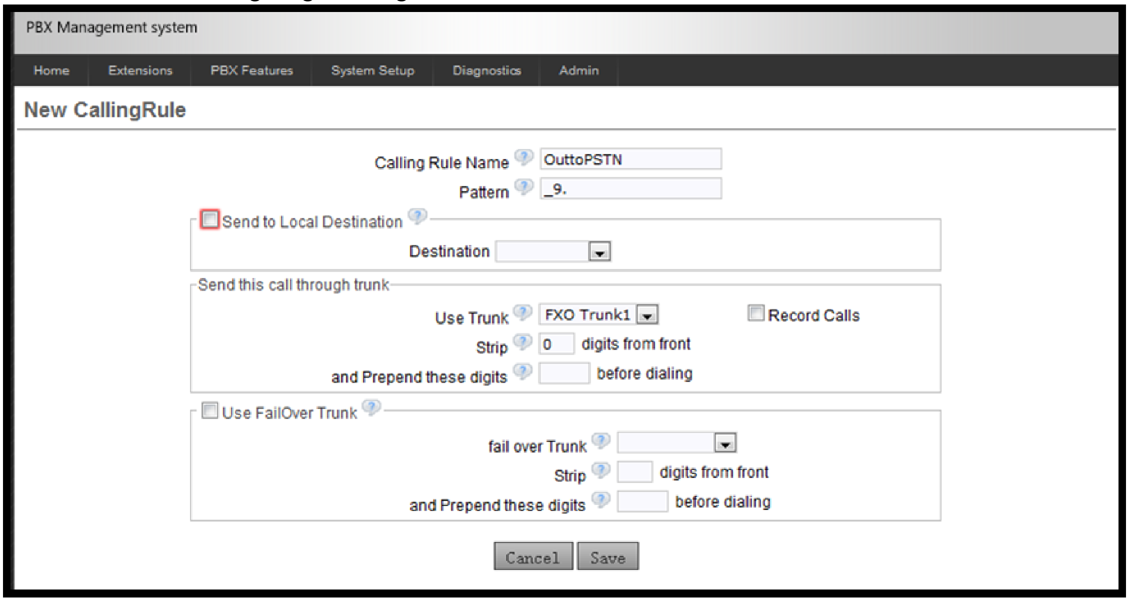
How to Communicate with Outside

In order to communicate with outside by IP PBX-02\04\08, users need an analog trunk, an outgoing calling rule, a dial plan, a incoming calling rule and a user. Here the writer will give the simple configuration steps which show how to make a call to outside.

First: Create an Analog Trunk
After logging into the web page of IP PBX-02\04\08, please click on Trunks Analog Trunks, Click "New Analog Trunk ", And click "Save" .
the writer configure an analog trunk like the following:



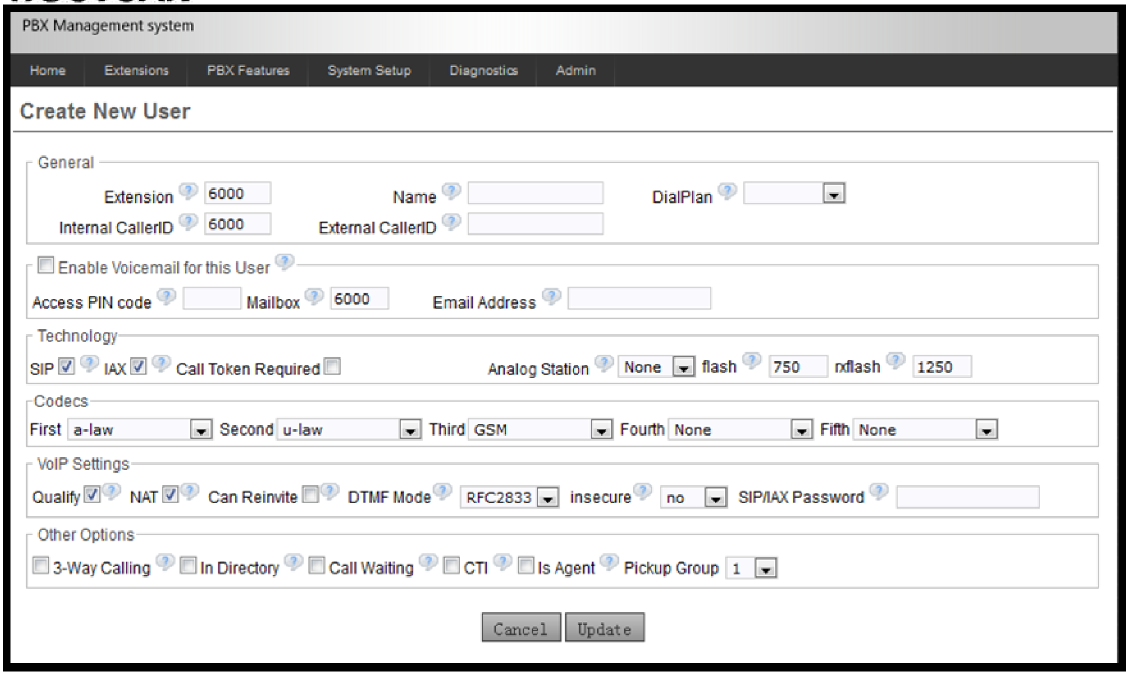
Next: Create an Outgoing Calling Rule



At last, please click on Save button, and click on Apply Changes button in up right corner of the main page.

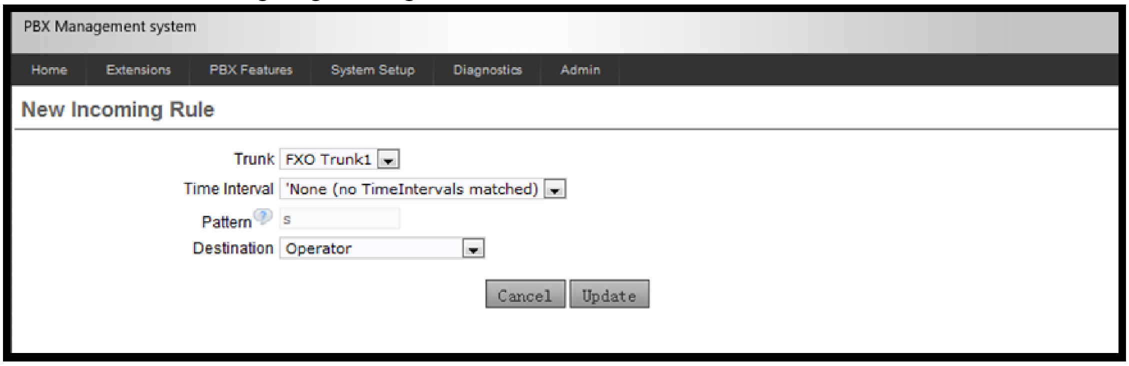
Next : Add the Rule to Dialplan
After logging into the web page of IP PBX-02\04\08, please click on Dial Plans Edit DialPlan1

Next: Create a User
After logging into the web page of IP PBX-02\04\08, please click on Users Create New User, the writer configure user 6000 like the following :



At last, please click on Update button, and click on Apply Changes button in up right corner of the main page.

Next : Create an Outgoing Calling Rule



At last, please click on Update button, and click on Apply Changes button in up right corner of the main page.

Here the users use the first channel. Then when the outside makes a incoming call, it will be sent to user 6000 through the first channel. Of course users can communicate with other use FXO/FXS by wireless.

For example:

The writer uses the channel 1 and the number is 158xxxxxxx2. Incoming Calling Rules be pointed to 6000. Then The writer can dial a mobile phone number with prefix 5, others can dial 158 xxxxxxx 2 to connect us.

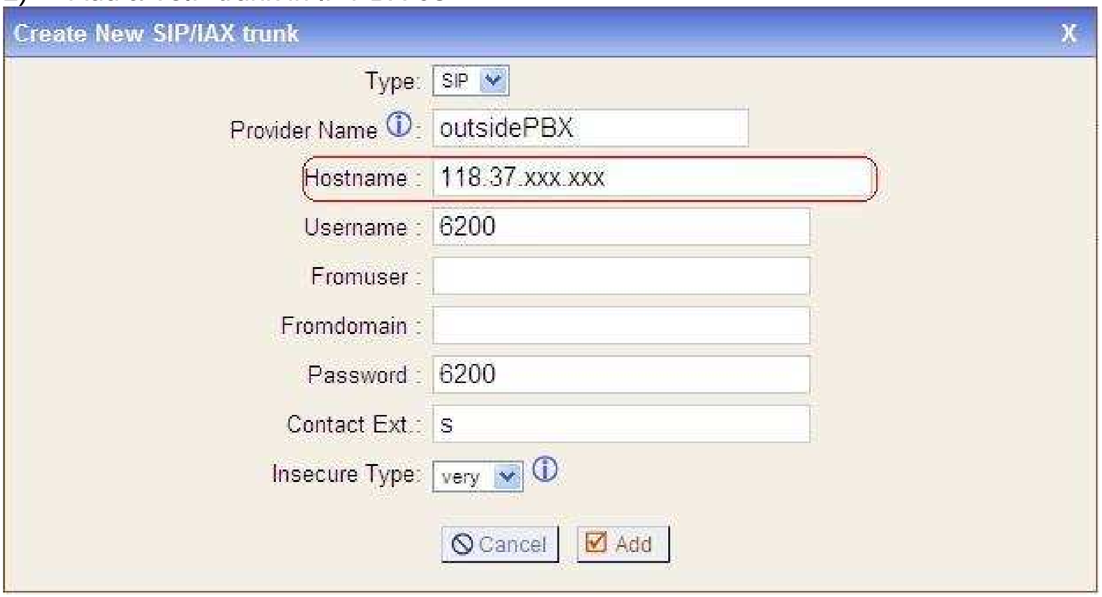
How to Call through VoIP Trunk

Call from IP PBX-08 to IP PBX-08

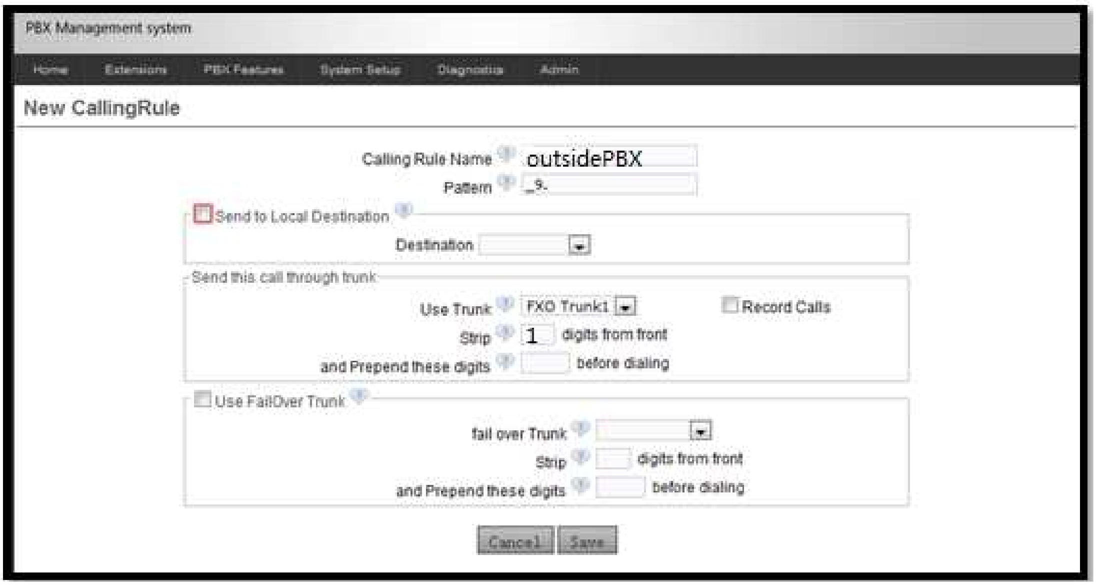
In order to call from IP PBX-08 to IP PBX-08, The writer will create a user in IP PBX-08 for the SIP/IAX trunk in IP PBX-08, create a SIP/IAX trunk, an outgoing call rule and a dial plan in IP PBX-08. But pay a attention that at the same time a port of the router where the IP PBX-08 in must be directed to the IP PBX-08.

1) Add an user 6200(it will be used as SIP trunk in IP PBX-08) in IP PBX-08, Then Add a user 6030 in IP PBX-08 for IP PHONE, the way is the same as adding 6001.

2) Add a VoIP trunk in IP PBX-08



3) Create an outgoing calling rule in IP PBX-08



4) Hook on the outgoing calling rules in dial plan in IP PBX-08

Now users can call from 6001 to 6030 by dialing 96030

Voicemail

Users can configure Voicemail in the option of Users, for example 6005 which the writer has configured in 3.319. Please click on Users Edit on 6001, users can see the configuration in the following picture, especially pay attention to the configuration in the red ellipse frame. Then when users want to listen to a message, they can dial 6750 or the Mailbox 6001.

How to realize the IVR

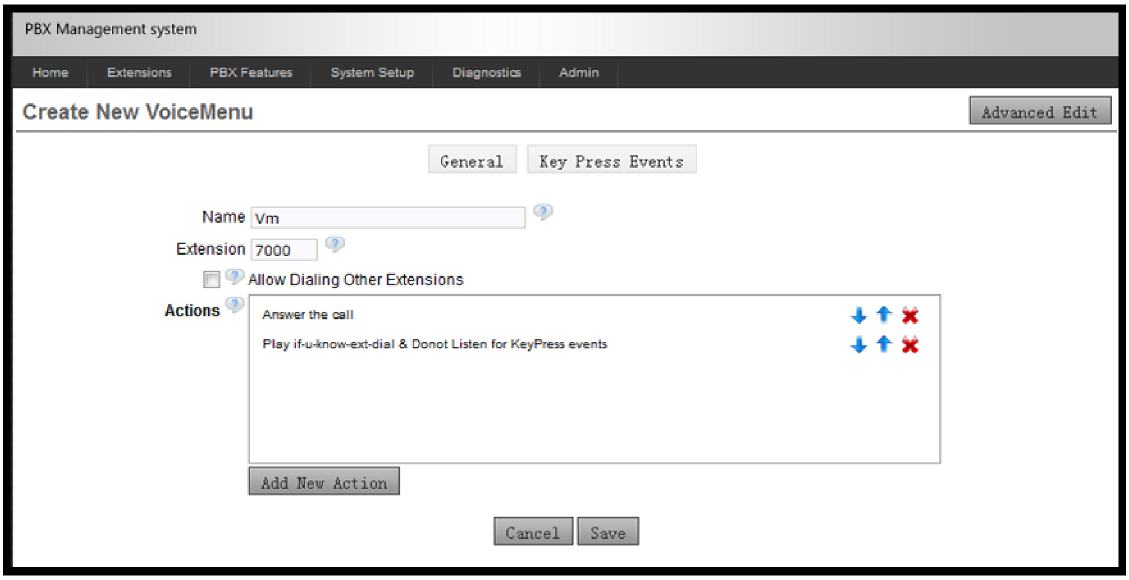
IVR is Interactive Voice Response. Voice Menus allow for more efficient routing of calls from incoming callers. Also known as IVR menus or Digital Receptionist.

Upload Voice Menu Prompts

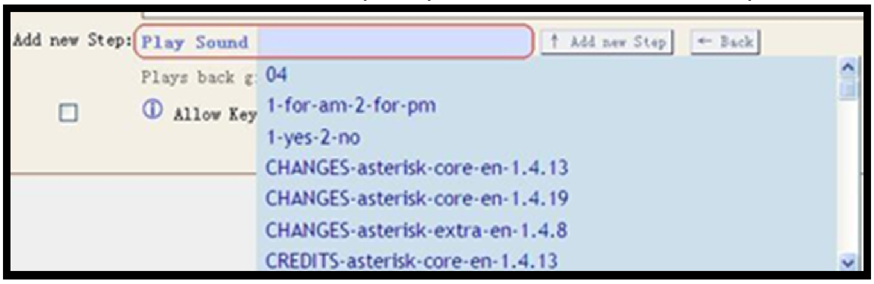
If users want to configure the IVR which they need, they must upload their voice prompt. Users can click on Voice Menu Prompts, users can see the screen like this screenshots:

Users can click the button of “Record a new Voice Menu prompt” to record a voice prompt, or users can click the button of “Upload a Voice Menu prompt” to upload their voice prompt.

Create Voice Menu



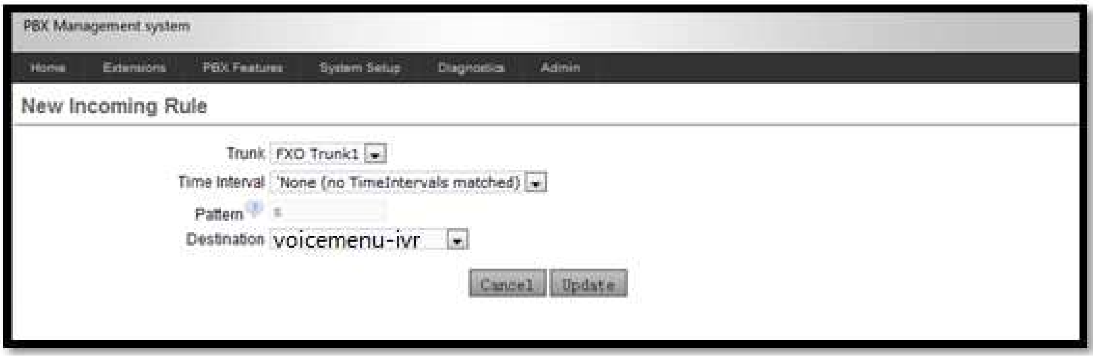
Selected the option “Background” on the “Add new step” then click the Add new step. Users can see the screen display like the following screenshots, then select their own voice prompt. Here the writer use the voice prompt named 04. Users can upload the voice prompt



Hook on the option : Allow KeyPress Envents, then users can configure the operation from “0” to “*”, which their need. Please click on save button, and click on Apply Changes button in up right corner of the main page. Here the writer configures that press “0” then call “6001”, press “1” then call “6002”, press “2” then call “6008”. Of course 6001, 6002, 6008 have registered.

Add Incoming Calling Rules

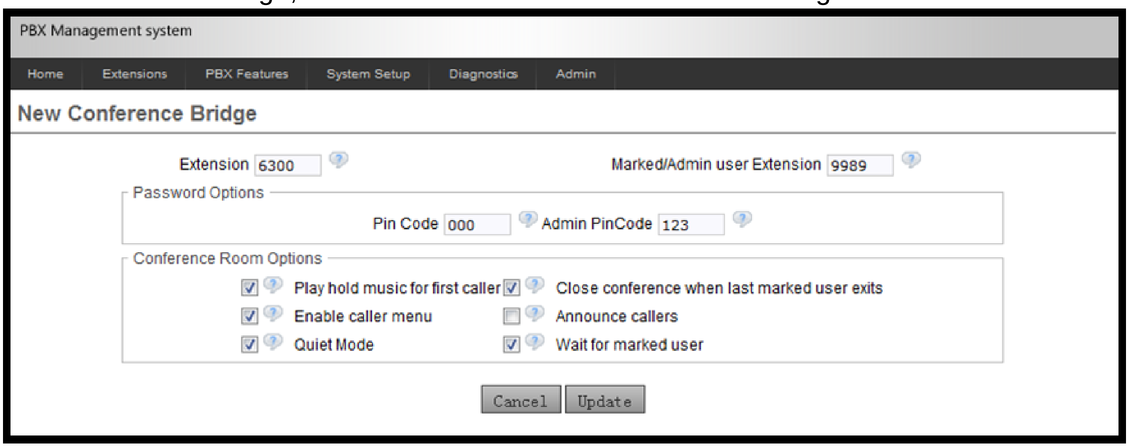
After configure the Voice Menu, users must configure the Incoming Calling Rules. Click Incoming Calling Rules New Incoming Calling Rules, users can configure it like this



Then when others call you through the analog1, they can here the IVR and do the operation which they need.

Conference

In order to realize the conference option, the users which will attend to the conference must have registered. Here the writer uses 6001, 6002, 6008. Now please click Conferencing New conference Bridge, users can see the screen like the following screenshots:

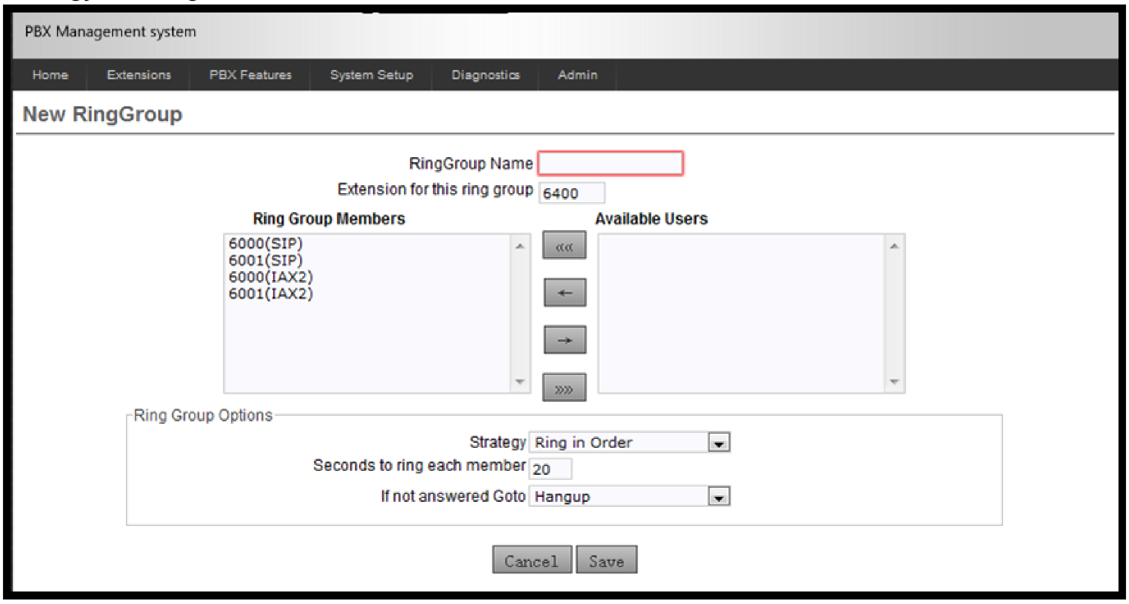


Then please click on Update button, and click on Apply Changes button in up right corner of the main page. Here the writer configures it like the screenshots above. Then 6001 dial 6300, and input Pin Code. Users can hear a voice promt and wait others, then you can hear the music. 6002 does the same operation. 6008 dial 9989 and input Admin PinCode. Now all the users are in the conference.

Ring Groups

Define Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Hunt groups. Users can click Ring Groups New Ring Group, then users can configure it like the following screenshots. Of course 6001,

6000 have registered. Then 6000 dial 6400, you can hear 6001, 6000 are ringing simultaneously. If users want the phones are ringing sequentially, they can configure the strategy as Ring in Order.



Agents

You need complete the following two steps when you need the function of Agents .

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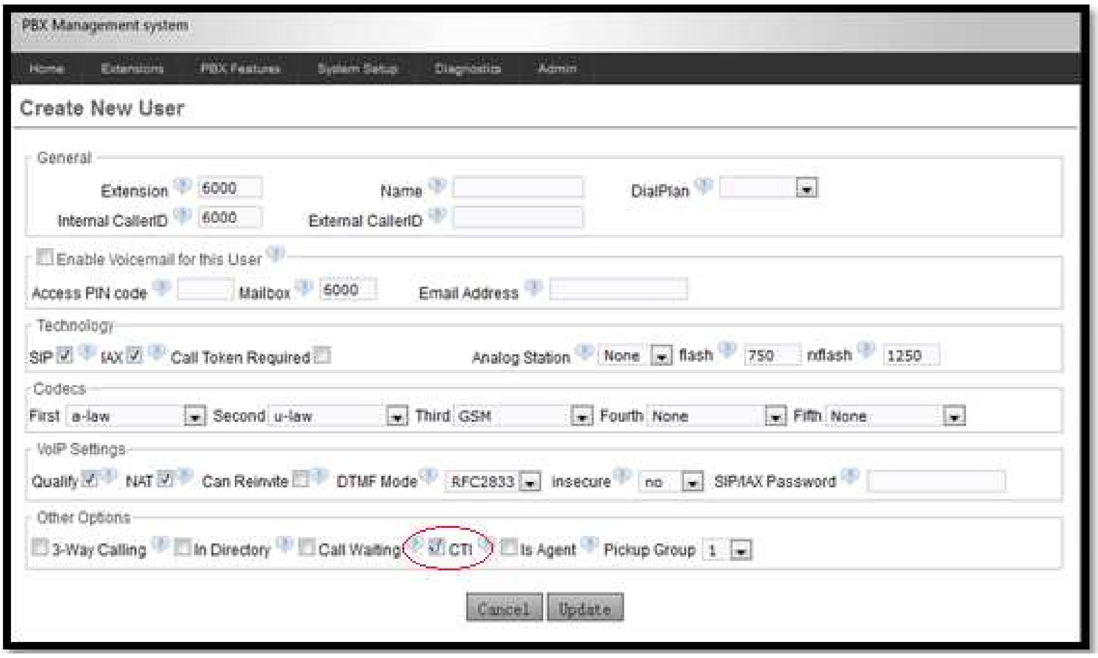
Create Users as Agents

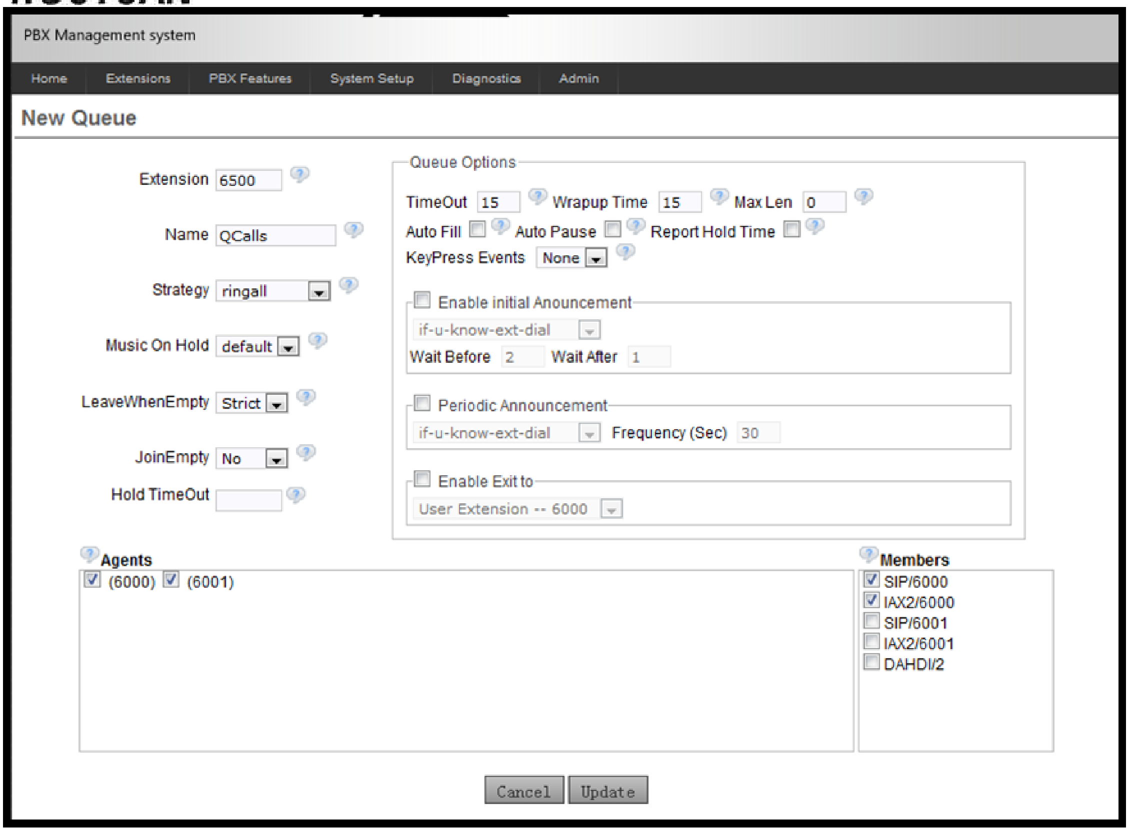


Like this I have also created 6002, 6008.Then you must click System Status, then you can see the following screenshots:

Click the button of "Login" so that all the Agents have logined. Then refresh the web, users can see the page that all the agents have logined like the following screenshots:

Create a Call Queue

Please click Call Queues   Create New Queue, then users can configure the options like this screenshots:



Then 6000(have registered) can call 6500, then 6001, 6000 are all ringing together.

Acronyms

VoIP: Voice over Internet Protocol
FXO: Foreign eXchange Office interface is the port that receives the analog line.
FXS: Foreign eXchange Subscriber interface is the port that actually delivers the analog line to the subscriber.
SIP: Session Initiation Protocol, SIP is a signalling protocol used for establishing sessions in an IP
network.
IAX: Inter-Asterisk Exchange Protocol, is a communications protocol for setting up interactive user sessions. IAX is similar to SIP.
RTP: Real-Time Transport Protocol, RTP is used to encapsulate VoIP data packets inside UDP packets. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services.
UDP: User Datagram Protocol, UDP is a communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that uses the Internet Protocol (IP).
TCP: Transmission Control Protocol , TCP is a set of rules (protocol) used along with the Internet Protocol (IP) to send data in the form of message units between computers over the Internet. SMTP: Simple Mail Transfer Protocol, SMTP is the de facto standard for electronic mail transport across the Internet.
TOS: Terms of service, the “ToS” or “TOS” are rules by which one must agree to abide by in order to use a service. Unless in violation of consumer protection laws, such terms are usually legally binding.
DTMF: Dual-tone multi-frequency, DTMF signaling is used for telephone signaling over the line in the voice-frequency band to the call switching center. The version of DTMF used for telephone tone dialing is known by the trademarked term Touch-Tone, and is standardised by ITU-T Recommendation Q.23. Other multi-frequency systems are used for signaling internal to the telephone network.
DHCP: Dynamic Host Configuration Protocol, DHCP is an auto configuration protocol used on IP networks. DHCP allows a computer to be configured automatically, eliminating the need for intervention by a network administrator. It also provides a central database for keeping track of computers that have been connected to the network. This prevents two computers from accidentally being configured with the same IP address.
NTP: Network Time Procotol, NTP is a protocol for synchronizing the clocks of computer systems over packet-switched, variable-latency data networks. It is designed particularly to resist the effects of variable latency by using a jitter buffer.
Vlan: Virtual Local Area Network, is a group of hosts with a common set of requirements that communicate as if they were attached to the same broadcast domain, regardless of their physical location. A VLAN has the same attributes as a physical LAN, but it allows for end stations to be grouped together even if they are not located on the same network switch. Network reconfiguration can be done through software instead of physically relocating devices.

HTTP: Hypertext Transfer Protocol, The HTTP is a networking protocol for distributed, collaborative, hypermedia information systems. HTTP is the foundation of data communication for the World Wide Web. HTTP functions as a request-response protocol in the client-server computing model. TFTP: Trivial File Transfer Protocol, TFTP is a file transfer protocol, with the functionality of a very basic form of File Transfer Protocol (FTP). TFTP could be implemented using a very small amount of memory. It was therefore useful for booting computers such as routers which did not have any data storage devices. It is still used to transfer small amounts of data between hosts on a network, such as IP Phone firmware or operating system images when a remote X Window System terminal or any other thin client boots from a network host or server. DNS: Domain Name System, The DNS is a distributed hierarchical naming system for computers, services, or any resource connected to the Internet or a private network. It associates various information with domain names assigned to each of the participants. Most importantly, it translates domain names meaningful to humans into the numerical (binary) identifiers associated with networking equipment for the purpose of locating and addressing these devices worldwide. MAC: Media Access Control address, The MAC is a unique identifier assigned to network adapters or network interface cards (NICs) usually by the manufacturer for identification. If assigned by the manufacturer, a MAC address usually encodes the manufacturer's registered identification number.

IPv4: Internet Protocol version 4, The IPv4 is the fourth revision in the development of the Internet Protocol (IP) and it is the first version of the protocol to be widely deployed.

NAT: Network Address Translation

DTMF: Dual Tone Multi Frequency

FXO/FXS: Global System for Mobile Communications

Glossary

Zaptel: Zaptel refers to Jim Dixon's open computer telephony hardware driver API. Zaptel drivers were first released for BSD and Jim's Tormenta series of DIY T1 interface cards. Digium later produced interface cards from Jim's designs and improved the Zaptel drivers on the Linux platform. Digium then added further drivers also following the Zaptel API for other telephony hardware.

Asterisk: Asterisk is a software implementation of a telephone private branch exchange (PBX) originally created in 1999 by Mark Spencer of Digium. Like any PBX, it allows attached telephones to make calls to one another, and to connect to other telephone services including the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services.

Voice Codec:

G.711 is a high bit rate (64 Kbps) ITU standard codec. It is the native language of the modern digital telephone network. There are two versions: A-law and U-law.

G.711 A-law is indigenous to the E1 standard used in the rest of the world. G.711 U-law is indigenous to the T1 standard used in North America and Japan. The difference is in the method the analog signal being sampled. In both schemes, the signal is not sampled linearly, but in a logarithmic fashion. A-law provides more dynamic range as opposed to U-law. The result is a less

'fuzzy' sound as sampling artifacts are better supressed.

Pick up: the ability to pull a ringing call to the phone you are currently on. There are two main types:-

a.Group call pickup, this allows you to collect a call from any ringing phone that is in the same pickup group as you, if there were more than one phone ringing then you would have no control over which call you collected.

b.Directed pickup, this allows you to pickup a call at a specific extension, maybe you're in another office and you hear a phone ringing and wonder if it's yours. You dial the pickup number and your extension, and the call will only transfer if it is your extension.

Group call pickup is typically invoked by dialing *8# or *8 from another phone in the call pickup group.

Syslog: Syslog is a standard for logging program messages. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices, which would otherwise be unable to communicate, a means to notify administrators of problems or performance.

Time Zone: A Time Zone is a region on Earth, more or less bounded by lines of longitude, that has a uniform, legally mandated standard time, usually referred to as the local time.